



Enterprise: Common

Solution Integration Guide for Communication Server 1000 Release 4.5/Business Communications Manager

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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 of product release. To check for updates to the latest documentation and software for Communication Server 1000 (CS 1000) and Business Communications Manager (BCM), click one of the following links:

For the...	Go to...
Latest CS 1000E software	Nortel page for CS 1000E software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=14261
Latest CS 1000M Cabinet/Chassis software	Nortel page for CS 1000M Cabinet/Chassis software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=12515
Latest CS 1000M Half Group/Single Group/Multi-Group software	Nortel page for CS 1000M Half Group/Single Group/Multi-Group software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=12516
Latest CS 1000S software	Nortel page for CS 1000S software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=12514
Latest BCM 200 software	Nortel page for BCM 200 software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=8236
Latest BCM 400 software	Nortel page for BCM 400 software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=17141

For the...	Go to...
Latest BCM50 software	Nortel page for BCM 400 software located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=SOFTWARE&resetFilter=1&poid=15181
Latest CS 1000E documentation	Nortel page for CS 1000E documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=14261
Latest CS 1000M Cabinet/Chassis documentation	Nortel page for CS 1000M Cabinet/Chassis documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=12515
Latest CS 1000M Half Group/Single Group/Multi-Group documentation	Nortel page for CS 1000M Half Group/Single Group/ Multi-Group documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=12516
Latest CS 1000S documentation	Nortel page for CS 1000S documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=12514
Latest BCM 200 documentation	Nortel page for BCM 200 documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=8236
Latest BCM 400 documentation	Nortel page for BCM 200 documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=17141
Latest BCM50 documentation	Nortel page for BCM 200 documentation located at: http://www130.nortelnetworks.com/go/main.jsp?cscat=DOCUMENTATION&resetFilter=1&poid=15181

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

About this document

This document describes the planning, configuration, and troubleshooting of the integration of the Business Communications Manager (BCM) with a Communication Server 1000 system. Integrate the CS 1000 and BCM systems when both systems have been installed and a baseline of operation has been achieved and tested.

The following systems and software releases are covered in this guide:

- Communication Server 1000 Release 4.5
- Business Communications Manager 200 Release 4.0
- Business Communications Manager 400 Release 4.0
- Business Communications Manager 50 Release 2.0

This document is intended to be a stand-alone guide, covering the prerequisites to and implementation of a successful CS 1000/BCM integration. A minimum skill set and level of understanding is assumed. References to other NTPs, engineering guides, or troubleshooting guides are made for informational purposes.

Audience

The intended audience for this document includes installation, planning, and maintenance personnel.

Related information

The following NTPs are referenced in this guide:

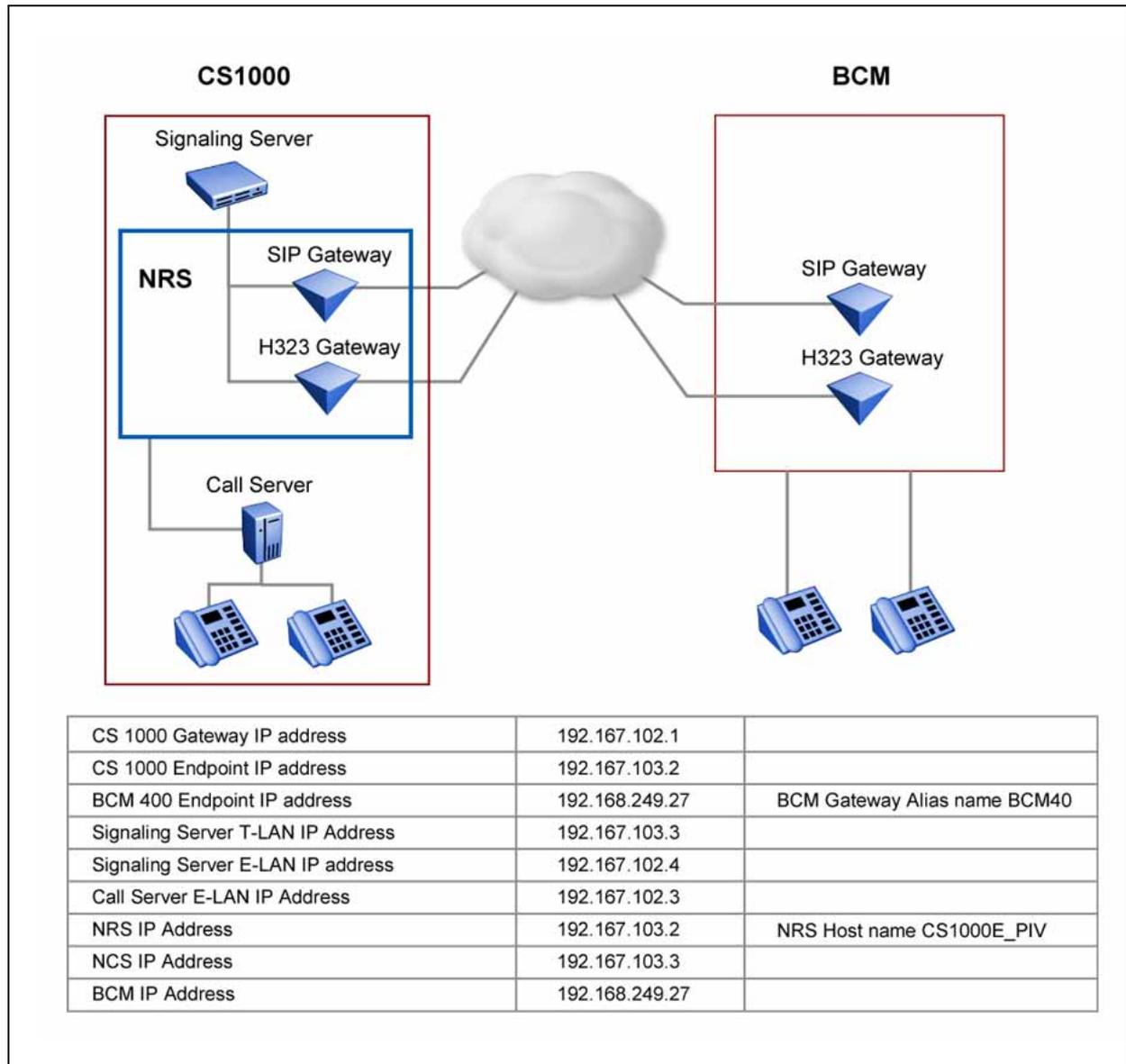
- *BCM 4.0 Device Configuration Guide* (N0060600)
- *BCM 4.0 Telephony Device Installation Guide* (N0060609)
- *Communication Server 1000E: Installation and Configuration* (553-3041-210)
- *Communication Server 1000M and Meridian 1: Large System Installation and Configuration* (553-3021-210)
- *Communication Server 1000M and Meridian 1: Small System Installation and Configuration* (553-3011-210)

- *Communication Server 1000S: Installation and Configuration (553-3031-210)*
- *Dialing Plans: Description (553-3001-183)*
- *IP Line Description, Installation, and Maintenance (553-3001-365)*
- *IP Peer Networking Installation and Configuration Guide (553-3001-213)*
- *Keycode Installation Guide (NN40010-301)*
- *Signaling Server: Installation and Configuration (553-3001-212)*

Overview

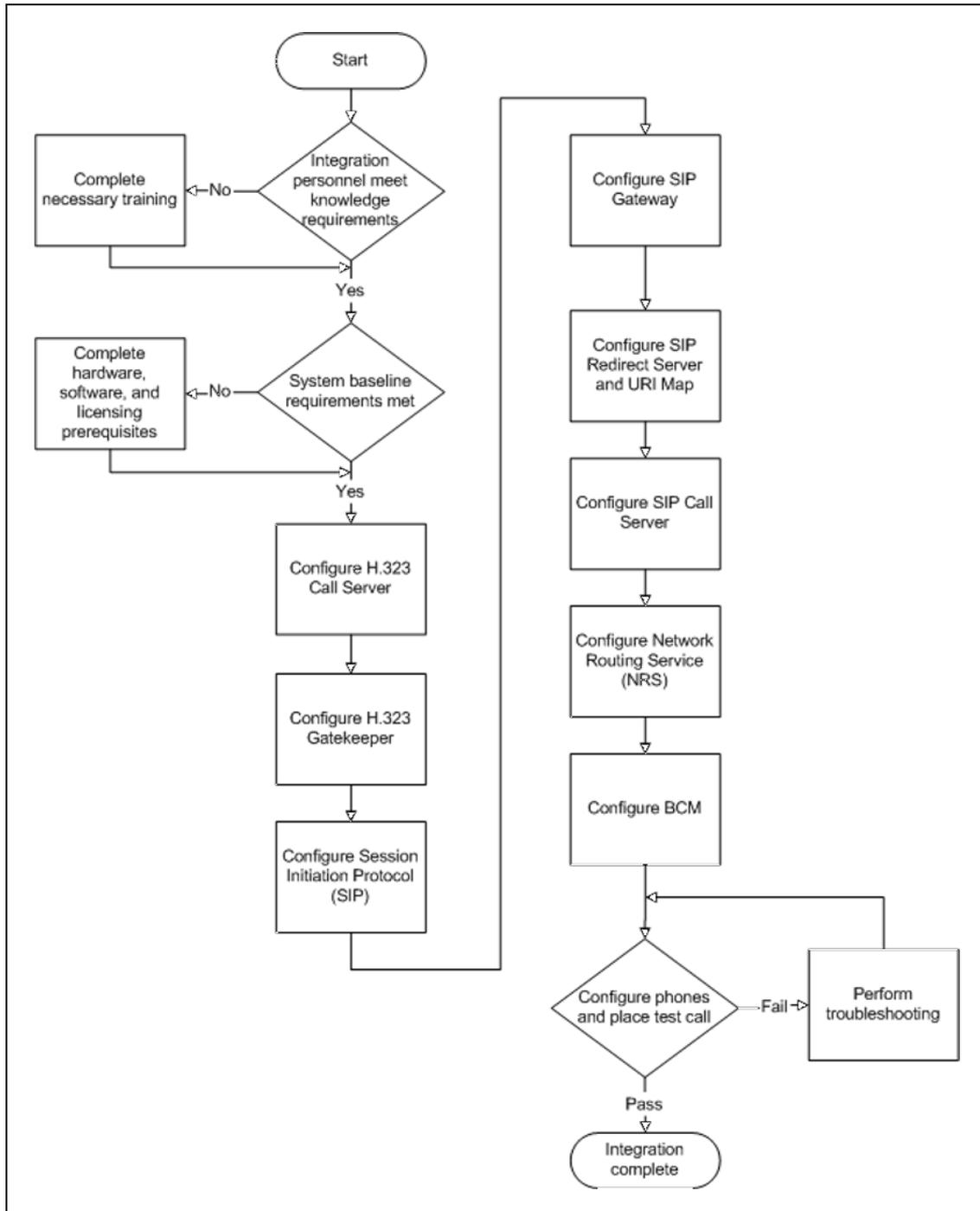
An example of a Communication Server 1000/Business Communications Manager (BCM) systems integration is shown in CS 1000/BCM architecture.

Figure 1
CS 1000/BCM architecture



CS 1000/BCM integration process shows the sequence of procedures you perform to integrate the CS 1000 and BCM systems.

Figure 2
CS 1000/BCM integration process



The tasks in the CS 1000/BCM systems integration process are listed in Task Completion Checklist. Use this checklist to implement the integration.

Table 1
Task Completion Checklist

	Task	Reference
	Configure the H.323 Call Server	<ol style="list-style-type: none"> 1. "Defining the customer to support ISDN" (page 31) 2. "Creating the virtual D-channel" (page 33) 3. "Configuring zones (LD 117)" (page 36) 4. "Creating the virtual route (LD 16)" (page 39) 5. "Creating the virtual trunks (LD 14)" (page 41) 6. "Creating the ESN data block for CDP" (page 43) 7. "Creating the Network Control Block (NCTL) for network access (LD 87)" (page 46) 8. "Creating the RLB for the virtual trunk route (LD 86)" (page 47) 9. "Creating the CDP steering codes (LD 87)" (page 49)
	Configure the H.323 Gatekeeper	"Configuring Element Manager" (page 53)
	Configure the SIP protocol	<ol style="list-style-type: none"> 1. "Enabling the SIP Virtual Trunk application" (page 57) 2. "Configuring the SIP Gateway" (page 58) 3. "Configuring the SIP Redirect Server and URI map" (page 60) 4. "Configuring IP networking for SIP" (page 62)
	Configure the SIP Call Server	<ol style="list-style-type: none"> 1. "Defining the customer to support ISDN" (page 62) 2. "Creating the virtual D-channel" (page 63) 3. "Configuring zones (LD 117)" (page 65) 4. "Creating the virtual route (LD 16)" (page 68) 5. "Creating the virtual trunks (LD 14)" (page 70) 6. "Creating the ESN data block for CDP" (page 72) 7. "Creating the Network Control Block (NCTL) for network access (LD 87)" (page 75) 8. "Creating the RLB for the virtual trunk route (LD 86)" (page 76) 9. "Creating the CDP steering codes (LD 87)" (page 78) 10. "Checking CODEC and QoS settings" (page 80)

	Task	Reference
	Configure NRS	<ol style="list-style-type: none"> 1. "Launching NRS Manager" (page 83) 2. "Verifying and adjusting system-wide settings" (page 85) 3. "Configuring the NRS server settings (H.323 Gatekeeper or SIP)" (page 87) 4. "Configuring the service domain" (page 89) 5. "Configuring the L1 domain (UDP)" (page 90) 6. "Configuring the L0 domain (CDP)" (page 93) 7. "Configuring Gateway endpoints" (page 96) 8. "Configuring routing entries" (page 100) 9. "Configuring collaborative servers" (page 102) 10. "Updating the database" (page 104) 11. "Checking the status of registered endpoints" (page 105) 12. "Checking the status of virtual D-channels" (page 106) 13. "Checking the status of virtual trunks" (page 107)
	Configure BCM	<p>BCM 200/400</p> <ol style="list-style-type: none"> 1. "Configuring incoming VoIP trunks" (page 111) 2. "Verifying system license and keycodes" (page 112) 3. "Configuring VoIP trunk media parameters" (page 113) 4. "Configuring local Gateway parameters" (page 116) 5. "Configuring VoIP lines" (page 121) 6. "Configuring target lines" (page 126) <p>BCM50</p> <ol style="list-style-type: none"> 1. "Configuring incoming VoIP trunks" (page 131) 2. "Verifying system license and keycodes" (page 132) 3. "Configuring VoIP trunk media parameters" (page 133) 4. "Configuring local Gateway parameters" (page 137) 5. "Configuring VoIP lines" (page 142)

	Task	Reference
		6. "Configuring target lines" (page 146)
	Test the integration	1. "Testing" (page 149) 2. "Troubleshooting" (page 151)

Prerequisites

Before you begin to integrate the Communication Server 1000 (CS 1000) and Business Communications Manager (BCM) systems, ensure that you complete the following prerequisites:

- Knowledge requirements
- Capturing integration parameters
- Establishing the system baseline

Knowledge requirements

The following knowledge and skills are required to implement a CS 1000/BCM systems integration:

- basic programming and provisioning skills for the CS 1000 system
- basic programming and provisioning skills for Network Routing Service (NRS)
- working knowledge of establishing dialing plans
- basic programming and provisioning skills for BCM systems
- working knowledge of various operating systems, including VxWorks, Unix, Linux, and Windows
- principles of Voice over IP (VoIP) protocols
- networking principles
- knowledge of core data components

Training

Nortel recommends that you complete product-specific training before you begin integrating the CS 1000 and BCM systems. Training includes course 6034C, "CS 1000 BCM Multi-site Integration", which deals specifically with the CS 1000/BCM integration and multi-site BCM integration processes. A complete list of courses is available at www.nortel.com

Capturing integration parameters

Integration parameters provides a list of parameters required to successfully complete the integration. Record these parameters during the initial planning phase of the integration.

Table 2
Integration parameters

Parameter	Value
User IDs and passwords	
SIP Gateway endpoint authentication password (must match the NRS password)	
IP addresses and URLs	
Gatekeeper IP address	
Alternate Gatekeeper IP address (optional)	
T-LAN IP address of the Signaling Server	
T-LAN IP address of the alternate Signaling Server	
Primary SIP proxy address	
Alternate SIP proxy address	
Primary NCS IP address	
Alternate NCS IP address)	
Static endpoint IP address (same as the Node IP address)	
Collaborative server IP address	
Names	
Service domain name in NRS	
SIP domain name (must be the same as the service domain name)	
SIP Gateway endpoint name (must match the NRS user ID)	
L0 domain name	
L1 domain name	
H.323 ID (preferable if it is the same as the one in the Primary Signaling Server)	
H.323 Gatekeeper alias name (default is the H.323 ID)	
Endpoint alias for BCM	
Read and write community names	
Miscellaneous	

Parameter	Value
Coordinated Dialing Plan steering codes	
SIP access port to use (port 5060 is recommended)	

Establishing the system baseline

To successfully integrate voice services, you must first establish the system baseline for the Call Server, Signaling Server, and Business Communications Manager (BCM) so that the systems are configured and working in a stand-alone environment.

Use Pre-integration checklist to complete system baselines prior to integration.

Table 3
Pre-integration checklist

	Task	Reference	Comments
	The Enterprise software package is purchased and installed, with appropriate licenses for virtual trunks, lines, and IP Phones as required.		
	The Network Numbering Plan is implemented.	<i>Dialing Plans: Description</i> (553- 3001-183)	Are you using a Uniform Dialing Plan (UDP) or a Coordinated Dialing Plan (CDP), or both? Are you also using a Group Dialing Plan (GDP), a North American Numbering Plan (NANP), or a Flexible Numbering Plan (FNP)?
	CS 1000 software is Release 4.5 or later.		To check the CS 1000 software release: Log on, enter LD 22 , and type PRT ISS . OR 1 Log on to Element Manager. 2 On the left navigation pane, select Home . The Home System View page appears.

22 Prerequisites

	Task	Reference	Comments
			<p>3 In the Call Server section, the software release is referred to as Release.</p>
	<p>Signaling Server software is Release 4.5 or later.</p>		<p>The Signaling Server software should be the most recent GA release compatible with your Call Server software version.</p> <p>To check the software release of the Signaling Server:</p> <p>1 Log on to Element Manager.</p> <p>2 On the left navigation pane, select Home. The Home System view page appears.</p> <p>3 Refer to the Signaling Server section for the Software Version.</p>
	<p>Basic installation, setup, and configuration of the Call Server components and the Signaling Server are complete.</p>	<p><i>Communication Server 1000M and Meridian 1: Small System Installation and Configuration (553-3011-210)</i></p> <p><i>Communication Server 1000M and Meridian 1: Large System Installation and Configuration (553-3021-210)</i></p> <p><i>Communication Server 1000S: Installation and Configuration (553-3031-210)</i></p> <p><i>Communication Server 1000E: Installation and Configuration (553-3041-210)</i></p> <p><i>Signaling Server: Installation and Configuration (553-3001-212)</i></p>	

	Task	Reference	Comments
	Primary, alternate, and fail-safe Network Routing Service (NRS) are configured at installation and initial setup of the Signaling Server.	<i>Signaling Server: Installation and Configuration</i> (553-3001-212)	The NRS requires IP telephony node configuration files. These files are installed and configured during the Signaling Server software installation as a basic configuration step.
	Digital Data Block configuration is complete in LD 73.	<i>IP Peer Networking Installation and Configuration Guide</i> (553-3001-213)	To configure a Digital Data Block: 1 Connect to the Call Server. 2 Enter LD 73 . 3 Enter NEW . 4 Enter DDB . 5 Press Enter to accept all defaults. 6 Perform a data dump.
	PTI or DTI trunks (DLOP) configuration is complete in LD 17.	<i>IP Peer Networking Installation and Configuration Guide</i> (553-3001-213)	To check PTI or DTI trunks: 1 Log on to Element Manager. 2 Select Routes and Trunks > Digital Trunk Interface . 3 Select Digital Trunk Interface Data Block . 4 Check that the configuration is complete.
	A basic node is configured in Element Manager.	<i>IP Line Description, Installation, and Maintenance</i> (553-3001-365)	This node information is updated through the integration process.

	Task	Reference	Comments
	Voice Gateway Media Card configuration is complete if IP to PSTN translation is required.		<p>To check that Media Gateway Cards are installed:</p> <ol style="list-style-type: none"> 1 Log on to Element Manager. 2 On the left side navigation pane, expand the IP Telephony tab. 3 Expand the Software tab. 4 Select Servers and Media Cards. The Servers and Media View page appears. 5 Select Open all Nodes. <p>Attention: The servers and Media Cards installed and configured are listed under each node. Any installed Voice Gateway Media Card is listed under Type.</p>
	H.323 Virtual Trunk package 399 is installed.		<p>To check that the package is loaded:</p> <ol style="list-style-type: none"> 1 Connect to the Call Server. 2 Log on to the Signaling Server. 3 Enter LD 22. 4 Enter PRT. 5 Enter PKG 399. 6 The package is loaded if you do not receive a "package is restricted" message.

	Task	Reference	Comments
	IPT is Release 3.0 or newer if you are using IP Trunk cards.		<p>To check that IPT Trunk cards are installed:</p> <ol style="list-style-type: none"> 1 Log on to Element Manager. 2 On the left navigation pane, expand the IP Telephony tab. 3 Expand the Software tab. 4 Select Servers and Media Cards. The Servers and Media View page appears. 5 Select Open all Nodes. <p>Attention: The servers and Media Cards installed and configured are listed under each node. Any installed IPT Trunk cards are listed under Type.</p>
	BCM configuration is complete and passing data traffic.		
	BCM networking hardware is installed for integration.		<p>To check the installed hardware:</p> <ol style="list-style-type: none"> 1 Log on to Element Manager. 2 Select the Administration tab. 3 Expand the General folder. 4 Select Hardware Inventory. 5 Select the PCI Cards tab. The cards installed in BCM are listed.

	Task	Reference	Comments
	PEC III Media Service Cards (MSC) are later.		<p>PECIII MSCs are required for T.38 Fax and IP telephony.</p> <p>To check the PEC hardware:</p> <ol style="list-style-type: none"> 1 Log on to Element Manager. 2 Select the Administration tab. 3 Expand the General folder. 4 Select Hardware Inventory. 5 Select the PCI Cards tab. 6 Select the MSC PCI card and scroll down to the Details for Card section.
	<p>BCM 200/400 is Release 4.0 or later.</p> <p>BCM50 is Release 2.0 or later.</p>		<p>To check the software version:</p> <ol style="list-style-type: none"> 1 Log on to Element Manager. 2 Select the Configuration tab. 3 Expand the System folder. 4 Select Identification.
	BCM 200/400 systems on the same network as the systems being integrated are Release 4.0 or later.		<p>To check the software version:</p> <ol style="list-style-type: none"> 1 Log on to Element Manager. 2 Select the Configuration tab. 3 Expand the System folder. 4 Select Identification.

	Task	Reference	Comments
	VoIP Gateway Trunk licensing is purchased and loaded on BCM.	<i>Keycode Installation Guide</i> (NN40010-301)	To check Feature Licenses: 1 Log on to Element Manager. 2 Select the Configuration tab. 3 Expand the System folder. 4 Select Keycode .
	IP Client licensing is purchased and loaded on BCM.	<i>Keycode Installation Guide</i> (NN40010-30)	To check Feature Licenses: 1 Log on to Element Manager. 2 Select the Configuration tab. 3 Expand the System folder. 4 Select Keycode .
	MCDN feature licensing is purchased and loaded on BCM.	<i>Keycode Installation Guide</i> (NN40010-30)	To check Feature Licenses: 1 Log on to Element Manager. 2 Select the Configuration tab. 3 Expand the System folder. 4 Select Keycode .

CS 1000 setup and IP Peer Networking configuration

Although you can configure the Communication Server 1000 and IP Peer Networking through overlays, the use of Element Manager and Network Routing Services (NRS) Manager are recommended. The Element Manager Web server resides on the Signaling Server and you can access it directly through a Web browser or by using Optivity Telephony Manager (OTM). You must configure NRS through Network Routing Service Manager (NRS Manager), which you can access only through the Element Manager.

Configure the Call Server through Element Manager in the following order:

- H.323 Call Server
- H.323 Gate Keeper
- Session Initiation Protocol (SIP)
 - SIP Protocol
 - SIP Gateway
 - SIP Redirect Server and URI Map
 - SIP Call Server
 - Network Routing Service in the NRS Manager
 - SIP addressing
 - SIP virtual trunking

CS 1000/IP Peer Networking configuration procedures

The sequence of CS 1000/IP Peer Networking configuration procedures is as follows:

- H.323 Call Server configuration
 - DCAM-6327328
 - DCAM-6327343
 - DCAM-6327347

- DCAM-6327351
- DCAM-6327354
- DCAM-6327360
- DCAM-6327357
- DCAM-6327364
- DCAM-6327367
- DCAM-6327370
- H.323 Gatekeeper configuration
 - Configuring Element Manager
- SIP protocol configuration
 - Enabling the SIP Virtual Trunk application
 - Configuring the SIP Gateway
 - Configuring the SIP Redirect Server and URI map
 - Defining the customer to support ISDN
 - Creating the virtual D-channel
 - Configuring zones (LD 117)
 - Creating the virtual route (LD 16)
 - Creating the virtual trunks (LD 14)
 - Creating the ESN data block for CDP
 - Creating the Network Control Block (NCTL) for network access (LD 87)
 - Creating the RLB for the virtual trunk route (LD 86)
 - Creating the CDP steering codes (LD 87)
 - Checking CODEC and QoS settings

H.323 Call Server configuration

The procedures in this section are as follows:

- DCAM-6327328
- DCAM-6327343
- DCAM-6327347
- DCAM-6327351
- DCAM-6327354

- DCAM-6327360
- DCAM-6327357
- DCAM-6327364
- DCAM-6327367
- DCAM-6327370

Defining the customer to support ISDN

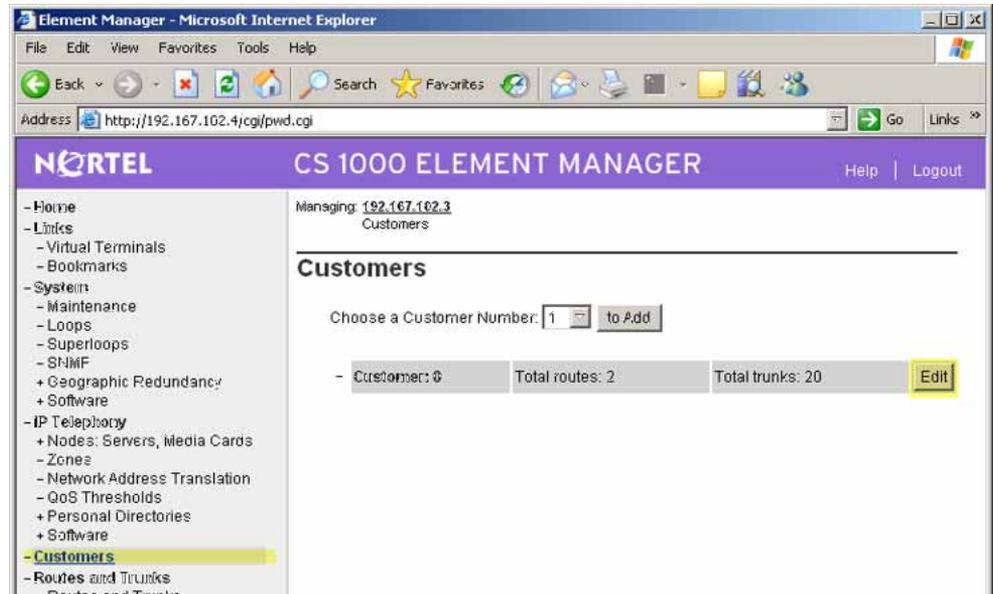
Complete the following procedure to define the customer to support ISDN.

Defining the customer to support ISDN

Step	Action
------	--------

- | | |
|---|--|
| 1 | Log on to Element Manager. |
| 2 | Select Customers .
See Figure 3 "Customers" (page 31). |

Figure 3
Customers



- | | |
|---|---|
| 3 | Click the Edit button.
The Customer Property Configuration page appears. See Customer Edit. |
|---|---|

Figure 4
Customer Property Configuration

Managing: 192.167.102.3
 Customers » Customer 0 Property Configuration

Customer 0 Property Configuration

- Basic Configuration

Input Description	Input Value
Customer Data Block (CDB) (TYPE)	CDB
Customer number (CUST)	0
ARB Attendant Billing number (ARBAT)	1111
ARB Listed Directory Number (ARLD)	111

Options (OPT)

• Flexible Feature Codes (FFC_DATA)
• Feature options (FTR_DATA)
• Listed Directory Number options (LDN_DATA)
• ISDN and ESN Networking options (HET_DATA)
• Night service options (NIT_DATA)

- Feature Packages

• Do Not Disturb Individual Package: 9

• End-to-End Simultaneous Package: 10

• Enhanced Night Service Package: 133

- Integrated Services Digital Network Package 145

Input Description	Input Value
Integrated Services Digital Network (ISDN)	<input checked="" type="checkbox"/>
Virtual Private Network Identifier (VPI)	0 Range: 1 - 16383
Private Network Identifier (PNI)	1 Range: 1 - 16383
Node DR (PRD_DN)	
Multi-location Business Group (MBG)	0 Range: 0 - 65535
Business Sub Group Consult-only (BSGC)	65535 Range: 0 - 65535
Prefix 1 (PK1)	123
Prefix 2 (PK2)	321
Home Number Plan Area code (HRPA)	Range: 200 - 999
Prefix for Central Office (HROG)	Range: 100 - 9999
Home location code (HLOC)	Range: 100 - 99999999
Local steering code (LSC)	
Calling Number Type (CKTP)	CLID feature displays the set's Prime DN (PDN) ▾
Redirection Count for ISDN calls (RCIT)	5
CLID information for incoming/outgoing calls (OCL)	No manipulation is done (NO) <input type="button" value="SC"/>

• Public Service Telephone Networks (PSTN)

- Flexible Services Package: 152 - Unequipped

• M3900 Product Enhancement Package: 386

- 4 Expand the **Feature Packages** heading.
- 5 Expand the **Integrated Services Digital Network Package 145** heading.

- 6 Select the **Integrated Services Digital Network (ISDN)** check box.
- 7 Click **Submit**.

—End—

Creating the virtual D-channel

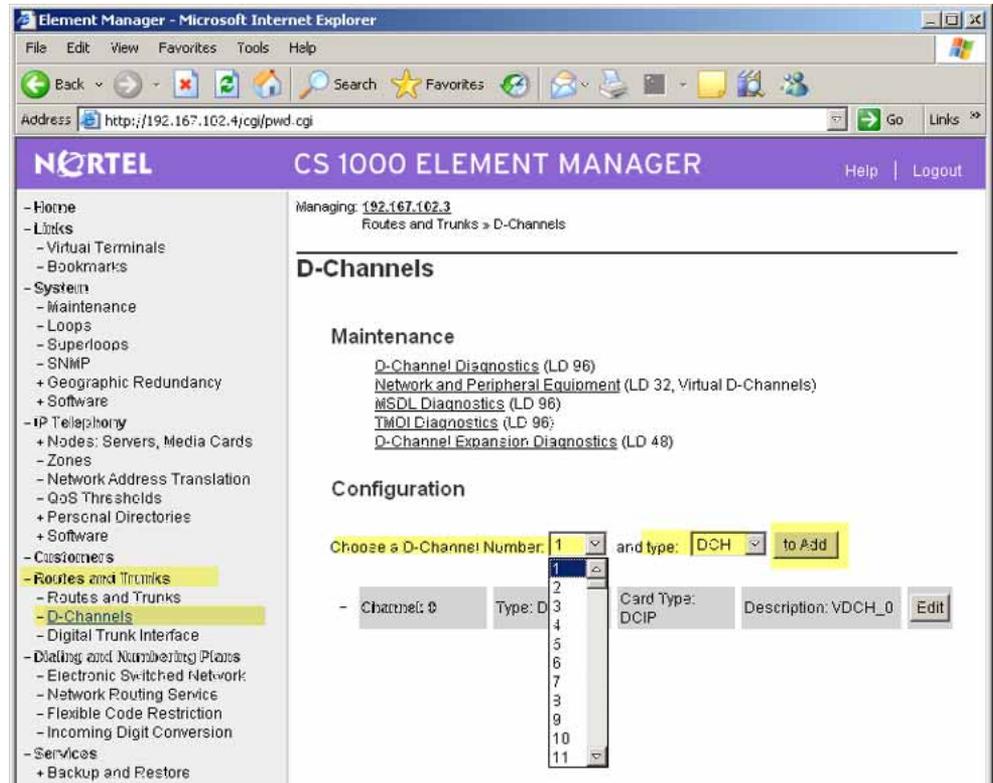
Perform the following procedure to create the virtual D-channel.

Creating the virtual D-channel

Step	Action
------	--------

- | | |
|---|--|
| 1 | Log on to Element Manager. |
| 2 | Select Routes and Trunks > D-Channels .
The D-Channels page appears. See D-Channels.
A message appears if a D-channel is not configured. Click OK . |

Figure 5
D-Channels



- 3 From the **Choose a D-Channel Number** menu, select the D-Channel number.

D-channels 0,1, and 2 are usually used or shared with other applications. It is recommended that you begin configuring virtual D-channels on channel 3.

- 4 From the **Type** menu, select the D-Channel type.
- 5 Click **to Add**.
The D-Channels Property Configuration page appears. See D-Channels Property Configuration.

Figure 6
D-Channels Property Configuration

Element Manager - Microsoft Internet Explorer
 Address: http://192.167.102.4/cgi/pwd.cgi

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 192.167.102.3
 Routes and Trunks > D-Channels > D-Channels 1 Property Configuration

D-Channels 1 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D-channel Card Type (CTYP)	D-Channel is over IP (DCIP)
Designator (DES)	VDCH_1
Recovery to Primary (RCVP)	<input type="checkbox"/>
User (USR)	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
Country (CMTRY)	ETS 300 = 102 basic protocol (ETS)
D-channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> more PRI
Secondary PRI2 loops (PRI2)	<input type="text"/>
Meridian 4 mode 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)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1600 Range: 0 - 4000

+ Basic options (BSCOPT)
+ Advanced options (ADVOPT)
+ Feature Packages

Submit Refresh Delete Cancel

- 6 For the **D Channel Card Type (CTYP)**, select **D-channel is over IP (DCIP)**.
- 7 For the **Designator (DES)**, type a meaningful name. The Designator must not contain spaces; use underscores instead. Make a note of the Designator in your records for future reference.

- 8 For **User (USR)**, select **Integrated Services Signaling Link Dedicated (ISLD)**.
- 9 For **Interface type for D-channel (IFC)**, select **Meridian Meridian1 (SL1)**.
- 10 Leave all other parameters as is and click **Submit**.
The new channel appears.

—End—

Configuring zones (LD 117)

Before you can configure the virtual routes and trunks, the following zones must be configured, in any order:

- Zone 1 = IP Phones zone (ZBRN = MO)
- Zone 2 = Voice Gateway Channels zone, which should be different from the IP Phones zone (ZBRN = VTRK)

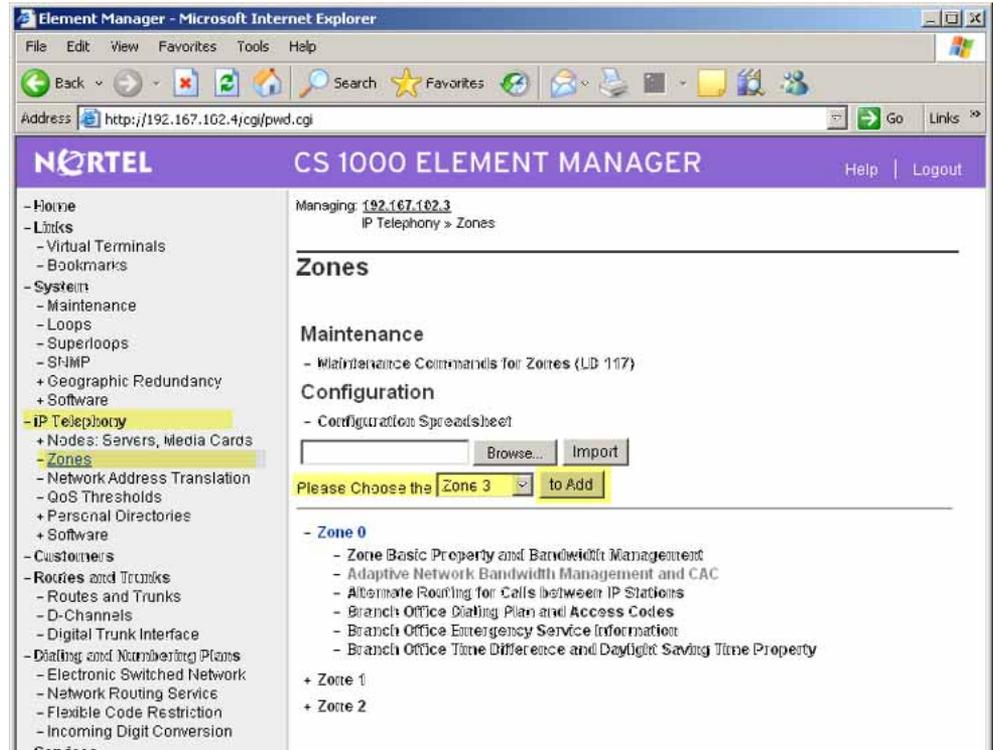
Ensure that enough bandwidth is allocated for the zones with the heaviest traffic.

Never use or configure zone 0.

Configuring zones (LD 117)

Step	Action
1	Log on to Element Manager.
2	Select IP Telephony > Zones . See Zones.

Figure 7
Zones



- 3 Select the **Zone** you wish to configure.
Configured zones appear in the list at the bottom of the page.
- 4 Click **to Add**.
The Zone Basic Property and Bandwidth Management page appears. See Zone Basic Property and Bandwidth Management.
- 5 After you click **to Add**, a message may appear prompting you to use the Zone Basic Property and Bandwidth Management Spreadsheet. Click **OK**.

Figure 8
Zone Basic Property and Bandwidth Management

The screenshot shows the 'Zone Basic Property and Bandwidth Management' configuration page in the Nortel CS 1000 Element Manager. The page is accessed via a web browser at the address http://192.167.102.4/cgi/pwd.cgi. The interface includes a navigation menu on the left and a main configuration area on the right. The configuration area is titled 'Zone Basic Property and Bandwidth Management' and contains the following fields:

Input Description	Input Value
Zone Number (ZONE):	3
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

At the bottom of the form, there are 'Submit' and 'Cancel' buttons.

- 6 Leave the default values for bandwidth and resource type as is.
- 7 Set the **Zone Intent (ZBRN)** as follows:
 - Zone 1 is for the IP Phones at the Main Office. Set Zone Intent (ZBRN) for Zone 1 to **MO**.
 - Zone 2 is for the Voice Gateway Channels. Set Zone Intent (ZBRN) for Zone 2 to **VTRK**.
- 8 For **Description (ZDES)**, type a meaningful description.
- 9 Click **Submit**.
- 10 Repeat this procedure for the second zone.

—End—

Creating the virtual route (LD 16)

Perform the following procedure to create the virtual route.

Creating the virtual route (LD 16)

Step	Action
------	--------

- | | |
|---|---|
| 1 | Log on to Element Manager. |
| 2 | Select Routes and Trunks > Routes and Trunks .
See Figure 9 "Routes and trunks" (page 39). |

Figure 9
Routes and trunks

The screenshot shows the 'Routes and Trunks' configuration page in the Nortel CS 1000 Element Manager. The interface includes a navigation menu on the left and a main content area. The main content area displays a summary of routes and trunks for a specific customer, with an 'Add route' button highlighted in yellow.

Customer	Total routes	Total trunks	Action
- Customer: 0	Total routes: 2	Total trunks: 20	Add route
- Route: 1	Type: TIE	Description: VTRK_H323	Edit Add trunk
- Trunk: 1	Total trunks: 10		
- Trunk: 1	TN: 096 0 02 00	Description: H323	Edit Multi-Del
- Trunk: 2	TN: 096 0 02 01	Description: H323	Edit
- Trunk: 3	TN: 096 0 02 02	Description: H323	Edit
- Trunk: 4	TN: 096 0 02 03	Description: H323	Edit
- Trunk: 5	TN: 096 0 02 04	Description: H323	Edit
- Trunk: 6	TN: 096 0 02 05	Description: H323	Edit
- Trunk: 7	TN: 096 0 02 06	Description: H323	Edit
- Trunk: 8	TN: 096 0 02 07	Description: H323	Edit
- Trunk: 9	TN: 096 0 02 08	Description: H323	Edit
- Trunk: 10	TN: 096 0 02 09	Description: H323	Edit
+ Route: 2	Type: TIE	Description: VTRK_SIF	Edit Add trunk

- | | |
|---|---|
| 3 | Click the Add route button.
The New Route Configuration page appears. See Route Property Configuration. |
|---|---|

Figure 10
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)	0
Designator field for trunk (DES)	VTRK_H323
Trunk Type (TKTP)	TIE trunk data block (TIE)
Incoming and Outgoing trunk (ICOG)	Incoming and Outgoing (IAO)
Access Code for the trunk route (ACOD)	1000
The route is for a virtual trunk route (VTRK)	<input checked="" type="checkbox"/>
- Zone for codec selection and bandwidth management (ZONE)	002 Range: 0 - 255
- Node ID of signaling server of this route (NODE)	9 Range: 0 - 9999
- Protocol ID for the route (PCID)	H323 (H323)
Integrated Services Digital Network option (ISDN)	<input checked="" type="checkbox"/>
- Mode of operation (MODE)	Route uses ISDN Signaling Link (ISLD)
- D channel number (DCH)	3 Range: 0 - 254
- Interface type for route (IFC)	Meridian #41 (SL1)
- Private Network Identifier (PNI)	0 Range: 0 - 32700
- Network Calling Name Allowed (NCNA)	<input checked="" type="checkbox"/>
- Network Call Redirection (NCRD)	<input type="checkbox"/>
- Recognition of DT12 ABCD/FALT signal for ISL (FALT)	<input type="checkbox"/>
- Channel Type (CHTY)	B-channel (BCH)
- Call Type for outgoing direct dialed TIE route (CTYP)	Coordinated Disting Plan (CDP)
- Insert ISN Access Code (INAC)	<input checked="" type="checkbox"/>
- Integrated Service Access Route (ISAR)	<input type="checkbox"/>
- Display of Access Prefix on CLID (DAPC)	<input type="checkbox"/>

+ Basic Route Options

+ Network Options

+ General Options

+ Advanced Configurations

Submit Cancel

- 4 Select the **Route Number (ROUT)**.
- 5 For **Designator field for trunk (DES)**, type a meaningful name.
- 6 For **Trunk Type (TKTP)**, select **TIE Trunk data block (TIE)**.
- 7 For **Incoming and Outgoing trunk (ICOG)**, select **Incoming and Outgoing (IAO)**.
- 8 Set the **Access Code for the trunk route (ACOD)**.
- 9 Select the **The route is for a virtual trunk route (VTRK)** check box.

- 10 Type the **Zone** number of the zone with the ZBRN set to Vtrk for the new route.
This value must match the values you configure in the Signaling Server.
- 11 Type the **Node ID of signaling server of this route (NODE)**.
This value must match the values you configure in the Signaling Server.
- 12 For **Protocol ID for the route (PCID)**, select **H323**.
- 13 Select the **Integrated Services Digital Network option (ISDN)** check box.
- 14 For **Mode of operation (MODE)**, select **Route uses ISDN Signaling Link (ISLD)**.
- 15 Select the virtual **D-Channel number (DCH)**.
- 16 For **Interface type for route (IFC)**, select **Meridian M1 (SL1)**.
- 17 Leave the **Call type for outgoing direct dialed TIE route (CTYP)** at the default value.
It is best to let NARS/BARS entries determine the NPI/TON for a number so that the route can be used for multiple call types.
- 18 Select the **Insert ESN Access Code (INAC)** check box.
- 19 Leave the other default values as is and click **Submit**.
The Routes and Trunks screen appears showing the created routes.

—End—

Creating the virtual trunks (LD 14)

The Virtual Trunk TNs that you configure in this procedure cannot overlap with the ones that you configure for IP Phones.

Create separate virtual routes for SIP and H.323. The SIP route is configured in the procedure Configuring IP networking for SIP.

Creating the virtual trunks (LD 14)

Step	Action
1	Log on to Element Manager.
2	Select Routes and Trunks > Routes and Trunks . See Figure 26 "Routes and trunks" (page 68) .
3	Expand the Customer heading.

- 4 Click **Add trunk** next to the route to which you wish to add the trunk. The New Trunk Configuration page appears. See New Trunk Configuration.

Figure 11
New Trunk Configuration

Element Manager - Microsoft Internet Explorer
Address: http://192.167.102.4/cgi/pwd.cgi

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 192.167.102.3
Routes and Trunks > Routes and Trunks > Customer 0, Route 1, New Trunk Configuration

Customer 0, Route 1, New Trunk Configuration

- Basic Configuration

Input Description	Input Value
Multiple trunk input number (MTINPUT)	2
Trunk data block (TYPE)	IP Trunk (IPTI)
Terminal Number (TN)	096 0 02 00
Designator field for trunk (DES)	H323
Extended Trunk (XTRK)	Virtual trunk (VTRK)
Customized number (CUST)	0
Route number, Member number (RTMB)	1
Card Density (CDEM)	Octal Density (8D)
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	1
Channel ID for this trunk (CHID)	10
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

+ Advanced Trunk Configurations

Submit Cancel

- 5 If you are configuring several trunks the same way, select the **Multiple trunk input number (MTINPUT)** (optional).
- 6 For **Trunk data block (TYPE)**, select **IP Trunk (IPTI)**.
- 7 Type the **Terminal Number (TN)** for the trunk.
- 8 For **Designator field for trunk (DES)**, type a meaningful value.
- 9 For **Extended Trunk (XTRK)**, select **Virtual trunk (VTRK)**.
- 10 Type the **Route number, Member number (RTMB)** for the trunk.
- 11 Set the values of **Start arrangement Incoming (STRI)** and **Start arrangement Outgoing (STRO)**.

Immediate (IMM) is recommended for both fields.

- 12 Type the **Channel ID for this trunk (CHID)**.
- 13 You can add a Class of Service (CLS) for all features that you wish. In a basic configuration, you can leave the CLS as is.
- 14 Select **Advanced Trunk Configurations** to display a list of advanced features.
- 15 Edit the necessary fields or accept the default values.
- 16 Click **Submit**.

—End—

Creating the ESN data block for CDP

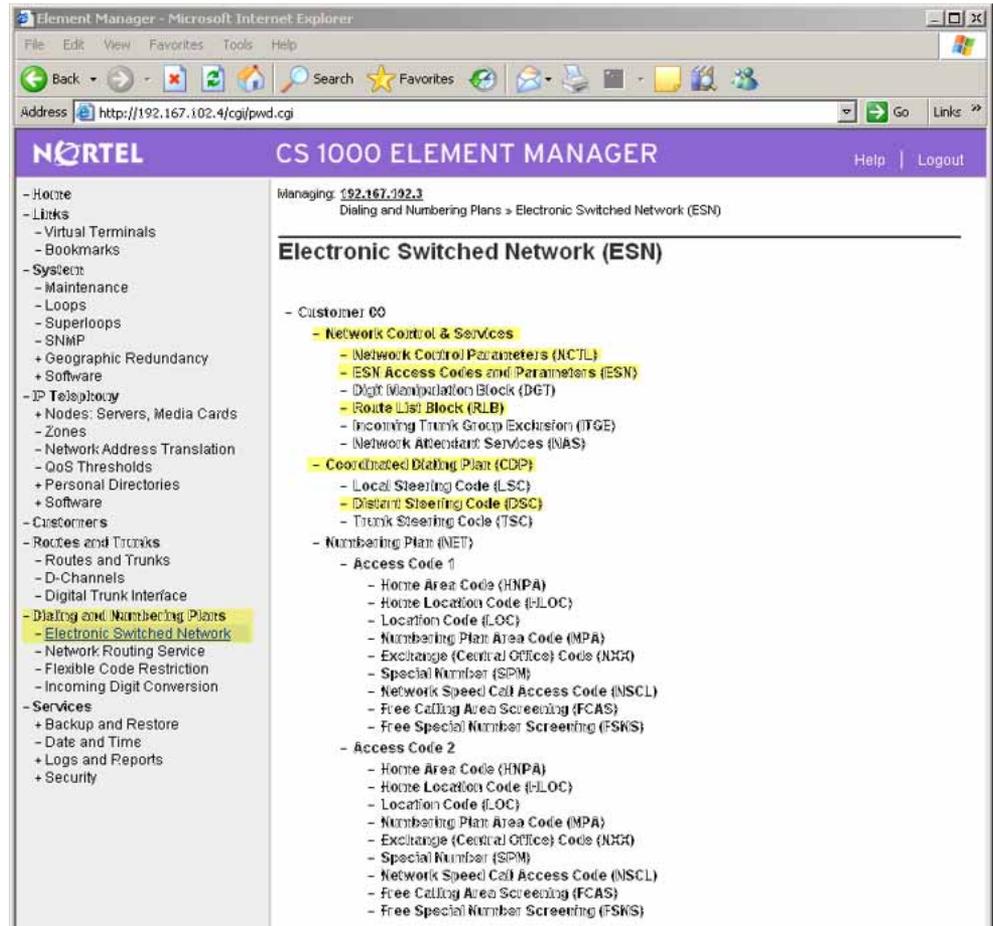
Perform the following procedure to create the ESN data block for CDP.

Creating the ESN data block for CDP

Step	Action
------	--------

- 1 Log on to Element Manager.
- 2 Select **Dialing and Numbering Plans > Electronic Switched Network**.
- 3 Expand the **Customer** heading.
See Electronic Switched Network.

Figure 12
Electronic Switched Network



- 4 Select **Network Control & Services > ESN Access Codes and Parameters (ESN)**.
- 5 A message appears if ESN data is not configured. Click **OK**. The ESN Access Codes and Basic Parameters page appears. See ESN Access Codes and Basic Parameters. If ESN data is configured on your switch, the fields on this page appear populated.

Figure 13
ESN Access Codes and Basic Parameters
ESN Access Codes and Basic Parameters

Input Description	Input Value
Maximum number of Digit Manipulation tables (MXDM):	100
Maximum number of Route Lists (MXRL):	100
Time of Day Schedules (TODS): (Items separated by a space)	0 00 00 23 59
Routing Controls (RTCL):	<input type="checkbox"/>
Click for Trunk Group Access Restrictions (TGAR):	<input type="checkbox"/>
NCOS Map (MXMAP): (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
Maximum number of Incoming Trunk Group exclusion tables (MXIX):	100
Maximum number of Free Calling area screening tables (MXFC):	100
Maximum number of Free Special number screening tables (MXFS):	100
One or two digit NARS/BARS Access Code 1 (AC1):	9
NARS/BARS Dial Tone after dialing AC1 or AC2 access codes (DLTN):	<input checked="" type="checkbox"/>
Expensive Route Warning Tone (ERWT):	<input checked="" type="checkbox"/>
Expensive Route Delay Time (ERDT):	6
Extended Time of Day schedule (ETOD):	
Maximum number of LOC codes (NARS only) (MXLC):	100
Maximum number of Special Common Carrier entities (MXSCC):	
One or two digit NARS Access Code 2 (AC2):	6
Coordinated Dialing Plan feature for this customer (CDP):	<input checked="" type="checkbox"/>
Maximum number of Steering Codes (MXSC):	120
Number of digits in CDP DN (DSC+DN or LSC+DN) (NCDP):	6
<input type="button" value="Submit"/> <input type="button" value="Refresh"/> <input type="button" value="Cancel"/>	

- 6 Edit the main parameters (**MXDM**, **MXRL**, **MXSD**, **MXIX**, **MXFC**, **MXFS** and **MXLC**) if required, or leave the default values as is.
- 7 Select the **Coordinated dialing Plan feature for this customer (CDP)** check box.
- 8 Set the value of the **Maximum number of Steering Codes (MXSC)**.
- 9 Set the value of the **Number of digits in CDP DN (DSC+DN or LSC+DN) (NCDP)**.

- 10 Click **Submit**.

—End—

Creating the Network Control Block (NCTL) for network access (LD 87)

Perform the following procedure to create the Network Control Block.

Creating the Network Control Block (NCTL) for network access (LD 87)

Step	Action
1	Log onto Element Manager.
2	Select Open Dialing and Numbering Plans > Electronic Switched Networks .
3	Expand the Customer tab. See Electronic Switched Network.
4	Select Network Control and Services > Network Control Parameter (NTCL) . A message appears if no network control data is configured. Click OK to configure new data.
5	Next to Network Control Basic Parameters, click the Edit tab. The Network Control Basic Parameters page appears. See Network Control Parameters.

Figure 14
Network Control Basic Parameters

Element Manager - Microsoft Internet Explorer
Address: http://192.167.104.4/cgi/pwd.cgi

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 192.167.104.3
Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Network Control & Services > Network Control Parameters > Network Class of Service Group

Network Class of Service Group

Input Description	Input Value
Network Class of Service group number (NCOS):	0
Maximum Precedence Level (MPL):	
Equal Access associated with this NCOS group (EAG):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0
Expansive Route Warning Tone (RWTA):	<input type="checkbox"/>
Network Speed Call access allowed (NSC):	<input type="checkbox"/>
Off-Hook Queuing eligibility (OHQ):	<input type="checkbox"/>
Call Back Queuing eligibility (CBQ):	<input type="checkbox"/>
Starting Priority in CBQ (SPRL):	0
Maximum Priority attainable in CBQ (MPRL):	0
Priority Promotion timer (PROW):	0
MLPP service domain class of service (MLSD):	000000
ARDL network route selection (ARDL):	Allowed from ALL (both initial and extended) route sets (A)

Submit Refresh Cancel

6 If required, type a **Network Class of Service group number (NCOS)**.

7 Click **Submit**.

—End—

Creating the RLB for the virtual trunk route (LD 86)

Perform the following procedure to create the RLB for the virtual trunk route.

Creating the RLB for the virtual trunk route (LD 86)

Step	Action
------	--------

1	Log on to Element Manager.
---	----------------------------

- 2 Select **Dialing and Numbering Plans > Electronic Switched Networks**.
- 3 Expand the **Customer** heading.
See Electronic Switched Network.
- 4 Select **Network Control and Services > Route List Blocks (RLB)**.
If route list blocks are not configured, the error message "Route List does not exist" appears. Click **OK**.
- 5 Type the **Route List Index** number.
- 6 Click to **Add**.
The Route List Block Configuration page appears. See Route List Block.

Figure 15
Route List Block

Input Description	Input Value
Route List Index (RLI):	2
Entry Number for the Route List (ENR):	0
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	1
Skips Conventional Signaling (SCNV):	<input type="checkbox"/>
Use Tone Detector (TDET):	<input type="checkbox"/>
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VMS):	<input type="checkbox"/>
Conversion to LDN (CNV):	<input type="checkbox"/>
Expensive Route (EXP):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0
Digit Manipulation Index (DMI):	0
ISL D-Channel Down Digit Manipulation Index (ISDM):	0
Free Calling Area Screening Index (FCI):	0
Free Special Number Screening Index (FSNI):	0
Business Network Extension Route (BNE):	<input type="checkbox"/>
Strategy on Congestion (SBOC):	Reroute All (RRA)
- 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):	<input type="checkbox"/>
Off-Hook Queuing Allowed (OHQ):	<input type="checkbox"/>
Call Back Queuing Allowed (CBQ):	<input type="checkbox"/>
Number of Alternate Routing Attempts (NALT):	5
Initial Set (ISET):	0
Set Minimum Facility Restriction Level (MIFRL):	
Overlap Length (OVL):	0

Submit Cancel

- 7 Select the **Route Number (ROUT)** you previously defined.
- 8 For **Strategy on Congestion (SBOC)**, select **Reroute All (RRA)**.
- 9 Accept the other defaults and click **Submit**.
The new Route List Block is generated. You can check the configuration by selecting Route List Block Index and Data Entry Index.

—End—

Creating the CDP steering codes (LD 87)

Perform the following procedure to create the CDP steering codes.

Creating the CDP steering codes (LD 87)

Step	Action
1	Log on to Element Manager.
2	Select Dialing and Numbering Plans > Electronic Switched Network .
3	Expand the Customer heading. See Electronic Switched Network.
4	Select Coordinated Dialing Plan (CDP) > Distant Steering Code List .
5	Enter the Distant Steering Code (DSC) . This is the DN range of other systems on the network. You can add more steering codes in this manner.
6	Click to Add . The Distant Steering Code page appears. See Distant Steering Code.

Figure 16
Distant Steering Code

The screenshot shows the 'Distant Steering Code' configuration page in the Nortel CS 1000 Element Manager. The browser address bar shows 'http://192.167.102.4/cgi/pwd.cgi'. The page title is 'Distant Steering Code'. The configuration area includes the following fields and options:

Input Description	Input Value
Distant Steering Code (DSC):	45
Flexible Length number of digits (FLEN):	0
Display (DSP):	Local Steering Code (LSC)
Remote Radio Paging Access (RRPA):	<input type="checkbox"/>
Route List to be accessed for trunk steering code (RLI):	1
Collect Call Blocking (CCBA):	<input type="checkbox"/>
maximum 7 digit NPA code allowed (NPA):	
maximum 7 digit NXX code allowed (NXX):	

Buttons at the bottom: Submit, Refresh, Delete, Cancel.

- 7 Check the populated fields.
- 8 Select a **Route list to be accessed for trunk steering code (RLI)**.
- 9 Click **Submit**.
- 10 Repeat steps 6 to 9 for all other call types on your network:
 - LOC (Location Code)
 - HLOC (Home Location Code)
 - NPA
 - HNPA (Home NPA)
 - SPN (Special Numbers)
 - NXX
- 11 This steering code is now defined. You can click the plus sign to view all the entered information.

—End—

Checking CODEC and QoS settings

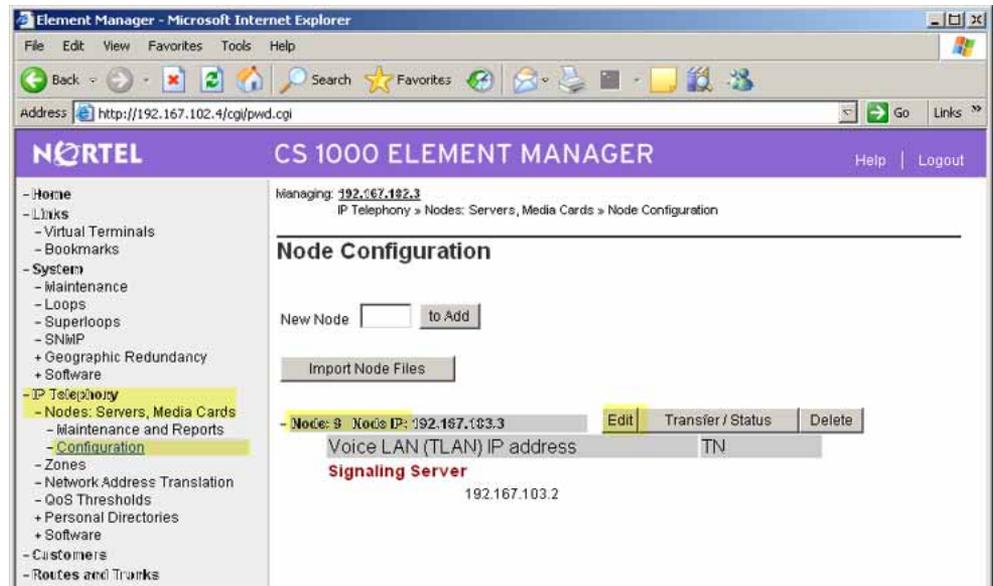
At this point, the Call Server configuration is complete. It is recommended that you check the CODEC and QoS settings.

Checking CODEC and QoS settings

Step	Action
------	--------

- | | |
|---|---|
| 1 | Log on to Element Manager. |
| 2 | Select IP Telephony Manager > Nodes: Servers, Media Cards > Configuration .
See Node Configuration. |

Figure 17
Node Configuration



- | | |
|---|---|
| 3 | Expand the Node heading. |
| 4 | Click Edit .
The Edit page appears. See Figure 18 "Node Editing" (page 52). |

Figure 18
Node Editing

Edit

Save and Transfer Cancel

- Mode Mode ID 9

Voice LAN (TLAN) Node IP address 192.167.103.3

Management LAN (ELAN) gateway IP address 192.167.102.1

Management LAN (ELAN) subnet mask 255.255.255.0

Voice LAN (TLAN) subnet mask 255.255.255.0

+ SNMP Add

- VGW and IP phone codec profile

Enable Echo canceler

Echo canceler 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 payload 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

- 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

+ LAN configuration

+ SNMP

+ H323 GW Settings

+ Firmware

+ SIP GW Settings

+ SIP URD Map

+ SIP CD Services

+ SIP CTI Services

+ Cards Add

+ Signaling Servers Add

Save and Transfer Cancel

- 5 Expand the **VGW and IP phone CODEC profile** heading and edit the fields as necessary.
- 6 Expand the **QoS** heading and edit the fields as necessary.

- 7 If you make configuration changes, click **Save and Transfer**; otherwise, click **Cancel**.

—End—

H.323 Gatekeeper Configuration

Configure the H.323 Gatekeeper in both Element Manager and the NRS. Configure the Element Manager before the NRS.

The sequence of H.323 Gatekeeper configuration procedures is as follows:

- Configuring Element Manager

Configuring Element Manager

Set the Primary and Alternate Gatekeeper IP addresses. This IP address is configured at each H.323 Gateway (Signaling Server).

Configuring Element Manager

Step	Action
1	Log on to Element Manager.
2	Select IP Telephony Manager > Nodes: Servers, Media Cards > Configuration .
3	Expand the Node ID heading. See Node Configuration.
4	Click Edit .
5	Expand the H.323 GW Settings heading. See H323 Gateway and Signaling Server.

Figure 19
H323 Gateway and Signaling Server

Edit

Save and Transfer Cancel

- Mode

Mode ID 9

Voice LAN (TLAN) Node IP address 192.167.103.3

Management LAN (ELAN) gateway IP address 192.167.102.1

Management LAN (ELAN) subnet mask 255.255.255.0

Voice LAN (TLAN) subnet mask 255.255.255.0

+ SNMP Add

- VQMF and SIP phone codec profile

Enable Echo canceler

Echo canceler 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 payload 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

- 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

+ LAN configuration

+ SNMP

+ H323 GW Settings

+ Firmware

+ SIP GW Settings

+ SIP URD Map

+ SIP CD Services

+ SIP CTI Services

+ Certs Add

+ Signaling Servers Add

Save and Transfer Cancel

- 6** Enter the **Primary Gatekeeper IP address**.
 This is the T-LAN IP address of the Signaling Server that runs the Gatekeeper application. This is not the same as the Node IP.

- 7 Enter the **Alternate Gatekeeper IP address** if you have an alternate on your system (optional).
- 8 Expand the **Signaling Servers** heading.
- 9 Expand the **Signaling Server Properties** heading.
- 10 Enter the **H323 ID**.
This should be the same as the Signaling Server that hosts the Primary H.323 Gatekeeper. Make a note of the H323 ID for the NRS configuration.
The host name does not need to be the same as the H323 ID, but it is recommended that both names are the same.
- 11 Select the **Enable Gatekeeper** check box.
- 12 Click **Save and Transfer**.

—End—

SIP protocol configuration

SIP addressing and Universal Resource Identifier

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 SIP:john.doe@companyabc.com. This URL is part of the Universal Resource Identifier (URI), which is used for SIP address resolution.

The user ID shows the composition of a SIP address. The SIP configuration of the Redirect Server is based on this address. The Domain name is the main address of a SIP server; you can compare it to a node ID. This is also the Service Domain that you enter in the NRS.

At the end of the URI, the user=phone section shows that this is the URI for a telephone user.

In SIP address resolution, the user-name field is parsed into name and phone context. Address lookup is based on digits, phone context, and domain name. This means that the telephone number undergoes SIP mapping. The Service Domain is divided into Level 0 (L0) and Level 1 (L1) domains. These domains are entered in the NRS.

SIP Gateway URI mapping and addressing

The SIP Gateway on the NRS provides the CS 1000 system with a direct trunking interface between the CS 1000 systems and a SIP domain.

The SIP Gateway has the following features:

- SIP User Agent (UA), which services one or more end-users in making/receiving SIP calls
- Signaling Gateway for all CS 1000 telephones (analog, digital, and IP Phones), which maps ISDN messages to and from SIP messages
- H.323 Gateway functionality
- MCDN, MWI partially mapped (mainly name display), partially tunneled
- ESN5, EPID tunneled

The SIP Gateway has the following functions:

- maps telephony numbers to and from SIP Uniform Resource Identifiers (URIs)
- performs client registration
- maps ISDN messages to and from SIP messages
- sets up the speech path between the desktop and SIP endpoints
- uses a standard SIP authentication security mechanism

The private numbering plans, public/unknown numbers and public/ special numbers also have explicit one-to-one mappings to SIP URI. They must be defined by preconfigured subdomain names. You must define the subdomain name on both the Gateway and proxy/registrar.

Configuration for the SIP protocol

Configuration for the SIP protocol is very similar to that of the Call Server.

The basic configuration for IP Peer Networking for SIP is performed in the following order:

1. Call Server in Element Manager
2. SIP Gateway
3. SIP Redirect Server and URI Map
4. Call Server
5. Network Routing Service in the NRS Manager
6. SIP addressing
7. SIP Virtual Trunking

If you select YES at the CRID prompt, a new line (SIP message) is added to the end of the Call Detail Record. This message shows the SIP addressing scheme.

The SIP Gateway also supports H.323 Gateway functionality so that both Gateways can interoperate. Also, you can point the virtual trunks, whether they are H.323 or SIP, to the same node ID on the signaling server. Then, the signaling server can perform the signaling for both protocols.

The procedures in this section are as follows:

- Enabling the SIP Virtual Trunk application
- Configuring the SIP Gateway
- Configuring the SIP Redirect Server and URI map
- Configuring IP networking for SIP

Enabling the SIP Virtual Trunk application

Perform the following procedure to enable SIP functionality in Element Manager. You must reboot the system during this procedure.

Enabling the SIP Virtual Trunk application

Step	Action
1	Log on to Element Manager.
2	Select IP Telephony > Nodes: Servers, Media Cards > Configuration . See Node Configuration.
3	Expand the Node heading.
4	Click Edit .
5	Expand the Signaling Servers heading.
6	Expand the Signaling Server Properties heading. See H323 Gateway and Signaling Server.
7	For Enable IP Peer Gateway (Virtual Trunk TPS) , select a SIP option (SIP only or H.323 and SIP).
8	Select the Enable SIP Proxy/Redirect Server check box.
9	Select the SIP Transport Protocol . TCP is the default. UDP means User Datagram Protocol in this instance.
10	Verify the Local SIP Port . The default is 5060.
11	Enter the SIP Domain Name .

The SIP Domain Name must be less than 128 characters in length. Valid characters are a-z, 0-9, period, hyphen, comma, and underscore.

This string builds all SIP messages and appears in the URI phone context. If you enable the SIP Gateway application, specify this field. This name must match the Service Domain name configured in NRS.

12 Enter the SIP Gateway Endpoint Name and Authentication Password.

These values must match the data in NRS. The SIP Gateway Endpoint Name becomes the Gateway's user ID. The user ID and password helps authenticate the Gateway with the MCS 5100 proxy server if you configure Converged Desktop.

13 Click Save and Transfer.

—End—

Configuring the SIP Gateway

Before you configure the SIP Gateway, check which route is configured as a SIP route in LD 16. You must configure this route later.

Configuring the SIP Gateway

Step	Action
------	--------

- | | |
|---|---|
| 1 | Log on to Element Manager. |
| 2 | Select IP Telephony > Nodes: Servers, Media Cards > Configuration .
See Node Configuration. |
| 3 | Expand the Node . |
| 4 | Click Edit . |
| 5 | Expand the SIP GW Settings heading.
See SIP GW settings. |

Figure 20
Edit SIP GW settings

CS 1000 ELEMENT MANAGER Help | Logout

Managing: [192.167.102.3](#)
 IP Telephony » Nodes: Servers, Media Cards » [Node Configuration](#) » IP Telephony: Node ID 9 » Edit

Edit

+ Node ○ ○ ○

-- Firmware

-- **SIP GW Settings**

Primary Proxy / Re-direct IP address

Primary Proxy / Re-direct IP Port

Primary Proxy Supports Registration

Primary CDS Proxy or Re-direct server flag

Secondary Proxy / Re-direct IP address

Secondary Proxy / Re-direct IP Port

Secondary Proxy Supports Registration

Secondary CDS Proxy or Re-direct server flag

CLID Parameters

Country Code

Area Code

Subscriber / Number of digits to strip

Subscriber / Prefix to insert

National / Number of digits to strip

National / Prefix to insert

◊ SIP URI Map ○ ○ ○

+ Signaling Servers

- 6 For the **Primary Proxy/Re-direct IP address**, type the T-LAN IP address of the Signaling Server that houses the Redirect Server.
- 7 For **Primary Proxy/Re-direct IP Port**, use the default port value of 5060.

You can use the T-LAN IP Address of the Alternate NRS in the Secondary Proxy/Re-direct IP address field.

- 8 Select the **Primary Proxy Supports Registration** check box.

—End—

Configuring the SIP Redirect Server and URI map

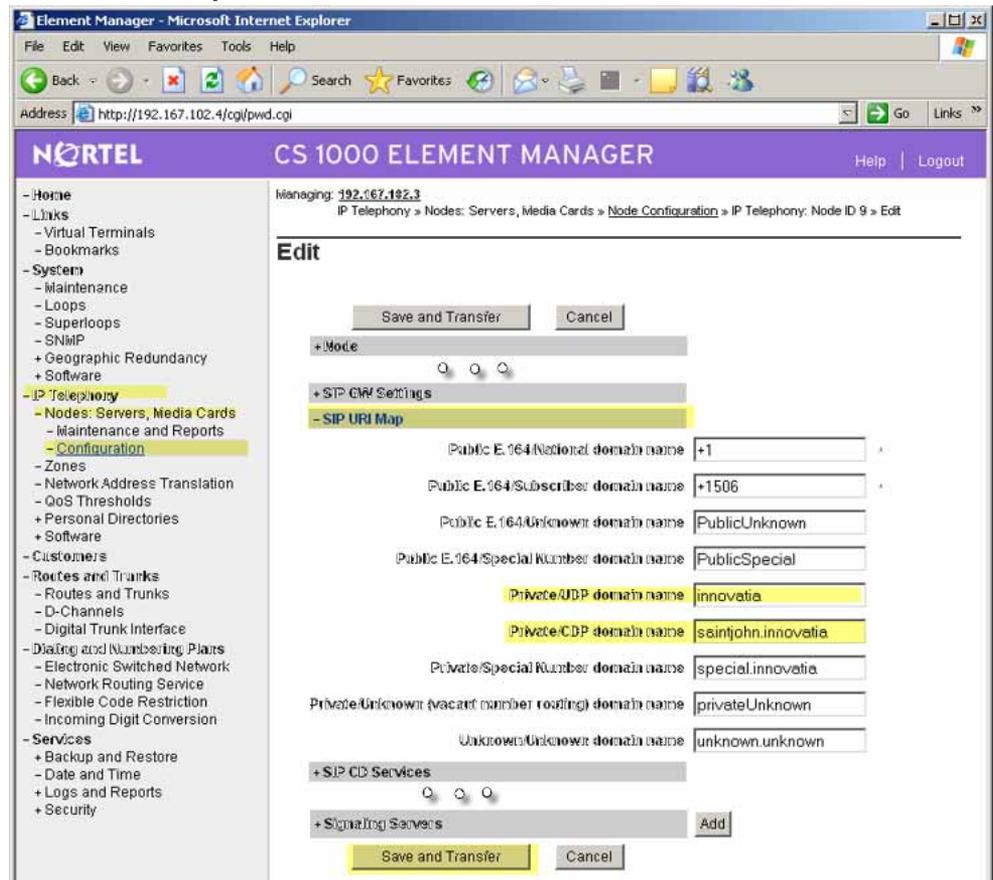
Use this procedure to configure your SIP numbering plan mapping. You can use this mapping to interpret TON/NPI numbers and map them to the associated context (to or from SIP). The TON/NPI field explicitly maps to the SIP phone-context attribute in the URI address.

Configuring the SIP Redirect Server and URI map

Step	Action
------	--------

- | | |
|---|---|
| 1 | Log on to Element Manager. |
| 2 | Select IP Telephony > Nodes:Servers, Media Cards > Configuration . |
| 3 | Expand the SIP URI Map heading.
See Edit SIP URI Map. |

Figure 21
Edit SIP URI Map



- 4 In the **Private/UDP domain name** field, type the L1 domain.
- 5 In the Private/CDP domain name field, type the L0 and L1 domains in the format <L0 domain.L1 domain>.
- 6 Enter the values for your SIP numbering plan in the appropriate fields.
- 7 Click **Save and Transfer**.
- 8 Click **OK** when the system is done transferring data and the successful transfer message appears.

—End—

Configuring IP networking for SIP

The IP Peer Networking configuration for SIP in the Call Server is similar to the H.323 configuration. Perform the following procedure for each Call Server in the IP Peer Network:

1. Define the customer to support ISDN (LD 15).
2. Create the virtual D-channel (LD 17).
3. Configure the zones (LD 117).
4. Create the virtual route (LD 16).
Enter **SIP** in the VTRK screen.
5. Create the virtual trunk.
6. Create the ESN data block for CDP (LD 86).
7. Create the Network Control (NCTL) block for network access (LD 87).
8. Create the RLB that uses the virtual trunk route (LD 86).
9. Create the CDP steering codes (LD 87).
10. If the system is configured for H.323, you do not need to configure steps 1, 2, 3, 6, and 7 again.

Defining the customer to support ISDN

Complete the following procedure to define the customer to support ISDN.

Defining the customer to support ISDN

Step	Action
1	Log on to Element Manager.
2	Select Customers . See Figure 3 "Customers" (page 31) .
3	Click Edit . A list of enabled feature packages appears.
4	Expand the Feature Packages heading.
5	Expand the Integrated Services Digital Network Package 145 heading.
6	Select the Integrated Services Digital Network (ISDN) check box.
7	Click Submit .

—End—

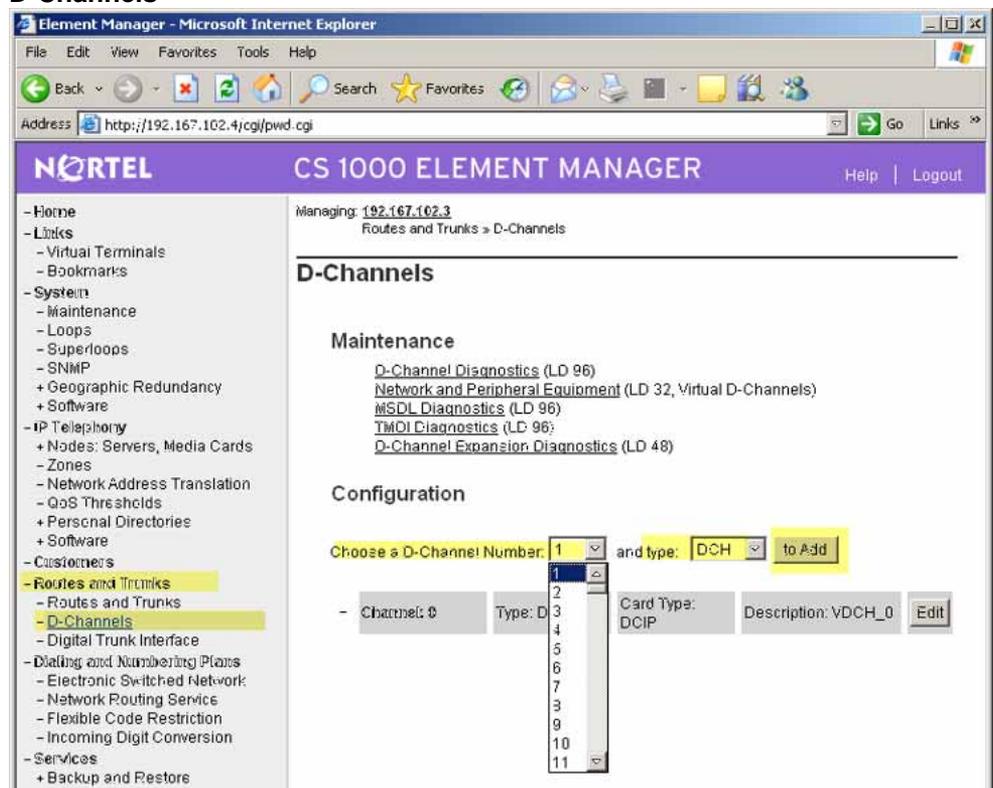
Creating the virtual D-channel

Perform the following procedure to create the virtual D-channel.

Creating the virtual D-channel

- | Step | Action |
|------|--|
| 1 | Log on to Element Manager. |
| 2 | Select Routes and Trunks > D-Channels .
The D-Channels page appears. See Figure 22 "D-Channels" (page 63).
A message appears if a D-channel is not configured. Click OK . |

Figure 22
D-Channels



- From the **Choose a D-Channel Number** menu, select the D-Channel number.
D-channels 0,1, and 2 are usually used or shared with other applications. It is recommended that you begin configuring virtual D-channels on channel 3.
- From the **Type** menu, select the D-Channel type.
- Click **to Add**.

The D-Channels Property Configuration page appears. See Figure 23 "D-Channels Property Configuration" (page 64).

Figure 23
D-Channels Property Configuration

Element Manager - Microsoft Internet Explorer
Address: http://192.167.102.4/cgi/pwd.cgi

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 192.167.102.3
Routes and Trunks > D-Channels > D-Channels 1 Property Configuration

D-Channels 1 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	D-Channel is over IP (DCIP)
Designator (DES)	VDCH_1
Recovery to Primary (RCVP)	<input type="checkbox"/>
User (USR)	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel (IC)	Meridian Meridian1 (SL1)
Country (CMTY)	ETS 300 = 102 basic protocol (ETS)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> <input type="button" value="more PRI"/>
Secondary PRI2 loops (PRI2)	<input type="text"/>
Meridian 1 mode type (MODE)	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)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1500 Range: 0 - 4000

+ Basic options (BSCOPT)

+ Advanced options (ADVOPT)

+ Feature Packages

- 6 For the **D Channel Card Type (CTYP)**, select **D-channel is over IP (DCIP)**.
- 7 For the **Designator (DES)**, type a meaningful name.

The Designator must not contain spaces; use underscores instead. Make a note of the Designator in your records for future reference.

- 8 For **User (USR)**, select **Integrated Services Signaling Link Dedicated (ISLD)**.
- 9 For **Interface type for D-channel (IFC)**, select **Meridian Meridian1 (SL1)**.
- 10 Leave all other parameters as is and click **Submit**. The new channel appears.

—End—

Configuring zones (LD 117)

Before you can configure the virtual routes and trunks, the following zones must be configured, in any order:

- Zone 1 = IP Phones zone (ZBRN = MO)
- Zone 2 = Voice Gateway Channels zone, which should be different from the IP Phones zone (ZBRN = VTRK)

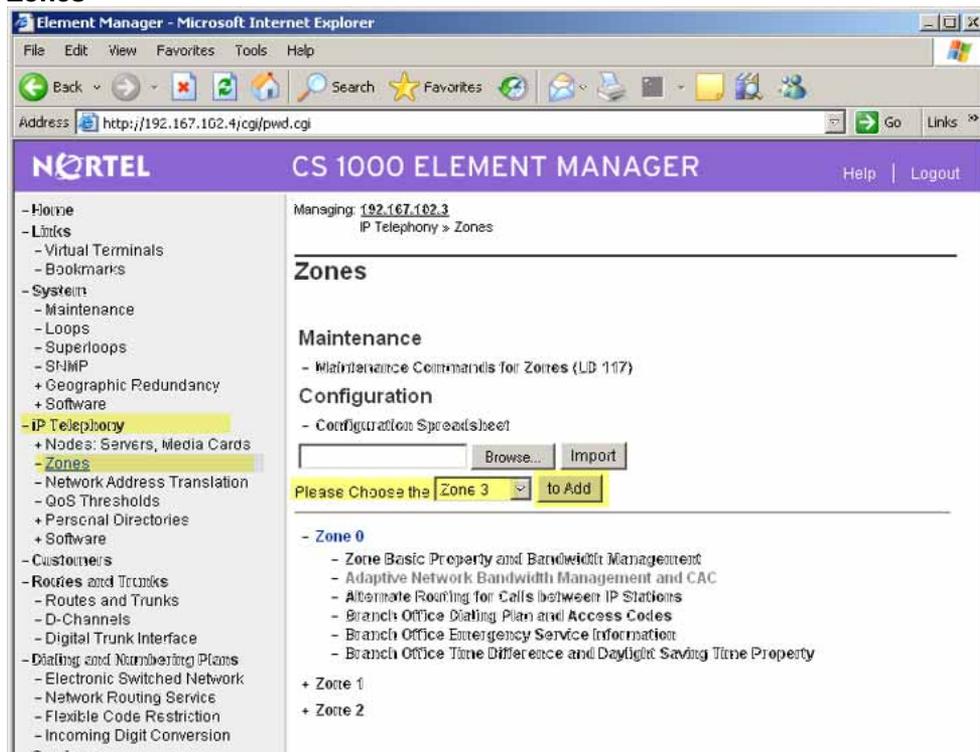
Ensure that enough bandwidth is allocated for the zones with the heaviest traffic.

Never use or configure zone 0.

Configuring zones (LD 117)

Step	Action
1	Log on to Element Manager.
2	Select IP Telephony > Zones . See Figure 24 "Zones" (page 66).

Figure 24
Zones



- 3 Select the **Zone** you wish to configure.
Configured zones appear in the list at the bottom of the page.
- 4 Click **to Add**.
The Zone Basic Property and Bandwidth Management page appears. See Figure 25 "Zone Basic Property and Bandwidth Management" (page 67).
- 5 After you click **to Add**, a message may appear prompting you to use the Zone Basic Property and Bandwidth Management Spreadsheet. Click **OK**.

Figure 25
Zone Basic Property and Bandwidth Management

The screenshot shows the 'Zone Basic Property and Bandwidth Management' configuration page in the Nortel CS 1000 Element Manager. The page is accessed via a web browser at the URL `http://192.167.102.4/cgi/pwd.cgi`. The interface includes a navigation menu on the left and a main content area with the following configuration fields:

Input Description	Input Value
Zone Number (ZONE):	3
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

At the bottom of the form, there are 'Submit' and 'Cancel' buttons.

- 6 Leave the default values for bandwidth and resource type as is.
- 7 Set the **Zone Intent (ZBRN)** as follows:
 - Zone 1 is for the IP Phones at the Main Office. Set Zone Intent (ZBRN) for Zone 1 to **MO**.
 - Zone 2 is for the Voice Gateway Channels. Set Zone Intent (ZBRN) for Zone 2 to **VTRK**.
- 8 For **Description (ZDES)**, type a meaningful description.
- 9 Click **Submit**.
- 10 Repeat this procedure for the second zone.

—End—

Creating the virtual route (LD 16)

Perform the following procedure to create the virtual route.

Creating the virtual route (LD 16)

Step	Action
------	--------

- | | |
|---|--|
| 1 | Log on to Element Manager. |
| 2 | Select Routes and Trunks > Routes and Trunks .
See Figure 26 "Routes and trunks" (page 68). |

Figure 26
Routes and trunks

The screenshot shows the 'Routes and Trunks' configuration page in the Nortel CS 1000 Element Manager. The page title is 'Routes and Trunks' and it shows the following data:

Customer	Total routes	Total trunks	Action
- Customer: 0	2	20	Add route
- Route: 1	Type: TIE	Description: VTRK_H323	Edit Add trunk
- Trunk: 1 -	Total trunks: 10		
10 - Trunk: 1	TN: 096 0 02 00	Description: H323	Edit Multi - Del
- Trunk: 2	TN: 096 0 02 01	Description: H323	Edit
- Trunk: 3	TN: 096 0 02 02	Description: H323	Edit
- Trunk: 4	TN: 096 0 02 03	Description: H323	Edit
- Trunk: 5	TN: 096 0 02 04	Description: H323	Edit
- Trunk: 6	TN: 096 0 02 05	Description: H323	Edit
- Trunk: 7	TN: 096 0 02 06	Description: H323	Edit
- Trunk: 8	TN: 096 0 02 07	Description: H323	Edit
- Trunk: 9	TN: 096 0 02 08	Description: H323	Edit
- Trunk: 10	TN: 096 0 02 09	Description: H323	Edit
+ Route: 2	Type: TIE	Description: VTRK_SIF	Edit Add trunk

- | | |
|---|--|
| 3 | Click the Add route button.
The New Route Configuration page appears. See Figure 27 "New Route Configuration" (page 69). |
|---|--|

Figure 27
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)	0
Designator field for trunk (DES)	VTRK_H323
Trunk Type (TKTP)	TIE trunk data block (TIE)
Incoming and Outgoing trunk (ICOG)	Incoming and Outgoing (IAO)
Access Code for the trunk route (ACOD)	1000
The route is for a virtual trunk route (VTRK)	<input checked="" type="checkbox"/>
- Zone for codec selection and bandwidth management (ZONE)	002 Range: 0 - 255
- Node ID of signaling server of this route (NODE)	9 Range: 0 - 9999
- Protocol ID for the route (PCID)	H323 (H323)
Integrated Services Digital Network option (ISDN)	<input checked="" type="checkbox"/>
- Mode of operation (MODE)	Route uses ISDN Signaling Link (ISL)
- D channel number (DCH)	3 Range: 0 - 254
- Interface type for route (IFC)	Meridian #41 (SL1)
- Private Network Identifier (PNI)	0 Range: 0 - 32700
- Network Calling Name Allowed (NCNA)	<input checked="" type="checkbox"/>
- Network Call Redirection (NCRD)	<input type="checkbox"/>
- Recognition of DT12 ABCD/FALT signal for ISL (FALT)	<input type="checkbox"/>
- Channel Type (CHTY)	B-channel (BCH)
- Call Type for outgoing direct dialed TIE route (CTYP)	Coordinated Disting Plan (CDP)
- Insert ISN Access Code (INAC)	<input checked="" type="checkbox"/>
- Integrated Service Access Route (ISAR)	<input type="checkbox"/>
- Display of Access Prefix on CLID (DAPC)	<input type="checkbox"/>

+ Basic Route Options

+ Network Options

+ General Options

+ Advanced Configurations

Submit Cancel

- 4 Select the **Route Number (ROUT)**.
- 5 For **Designator field for trunk (DES)**, type a meaningful name.
- 6 For **Trunk Type (TKTP)**, select **TIE Trunk data block (TIE)**.
- 7 For **Incoming and Outgoing trunk (ICOG)**, select **Incoming and Outgoing (IAO)**.
- 8 For **Access Code for the trunk route (ACOD)**, select an unused number.

- 9 Select the **The route is for a virtual trunk route (VTRK)** check box.
- 10 Type the **zone number**.
This value must match the value you configure in the Signaling Server.
- 11 Type the **Node ID of signaling server of this route (NODE)**.
This value must match the value you configure in the Signaling Server.
- 12 For **Protocol ID for the route (PCID)**, select **SIP**.
- 13 Select the **Integrated Services Digital Network option (ISDN)** check box.
- 14 For **Mode of operation (MODE)**, select **Route uses ISDN Signaling Link (ISLD)**.
- 15 Select the virtual **D-Channel number (DCH)**.
- 16 For **Interface type for route (IFC)**, select **Meridian M1 (SL1)**.
- 17 Leave the **Call type for outgoing direct dialed TIE route (CTYP)** at the default value.
It is best to let NARS/BARS entries determine the NPI/TON for a number so that the route can be used for multiple call types.
- 18 Select the **Insert ESN Access Code (INAC)** check box.
- 19 Leave the other default values as is and click **Submit**.
The Routes and Trunks screen appears showing the created routes.

—End—

Creating the virtual trunks (LD 14)

Create separate virtual routes for SIP and H.323. The H.323 route is configured in the procedure [Creating the virtual trunks \(LD 14\)](#).

Creating the virtual trunks (LD 14)

Step	Action
1	Log on to Element Manager.
2	Select Routes and Trunks > Routes and Trunks . See Figure 26 "Routes and trunks" (page 68) .
3	Expand the Customer heading.
4	Click Add trunk next to the route to which you wish to add the trunk.

The New Trunk Configuration page appears. See Figure 28 "New Trunk Configuration" (page 71).

Figure 28
New Trunk Configuration

Input Description	Input Value
Multiple trunk input number (MTINPUT)	2
Trunk data block (TYPE)	IP Trunk (IPTI)
Terminal Number (TN)	096 0 02 00
Designator field for trunk (DES)	H323
Extended Trunk (XTRK)	Virtual trunk (VTRK)
Customer number (CUST)	0
Route number, Member number (RTMB)	1 1
Card Density (CDEN)	Octal Density (8D)
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	1
Channel ID for this trunk (CHID)	10
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

- 5 If you are configuring several trunks the same way, select the **Multiple trunk input number (MTINPUT)** (optional).
- 6 For **Trunk data block (TYPE)**, select **IP Trunk (IPTI)**.
- 7 Type the **Terminal Number (TN)** for the trunk.
- 8 For **Designator field for trunk (DES)**, type a meaningful value.
- 9 For **Extended Trunk (XTRK)**, select **Virtual trunk (VTRK)**.
- 10 Type the **Route number, Member number (RTMB)** for the trunk.
- 11 Set the values of **Start arrangement Incoming (STRI)** and **Start arrangement Outgoing (STRO)**. Immediate (IMM) is recommended for both fields.

- 12 Type the **Channel ID for this trunk (CHID)**.
- 13 You can add a Class of Service (CLS) for all features that you wish. In a basic configuration, you can leave the CLS as is.
- 14 Select **Advanced Trunk Configurations** to display a list of advanced features.
- 15 Edit the necessary fields or accept the default values.
- 16 Click **Submit**.

—End—

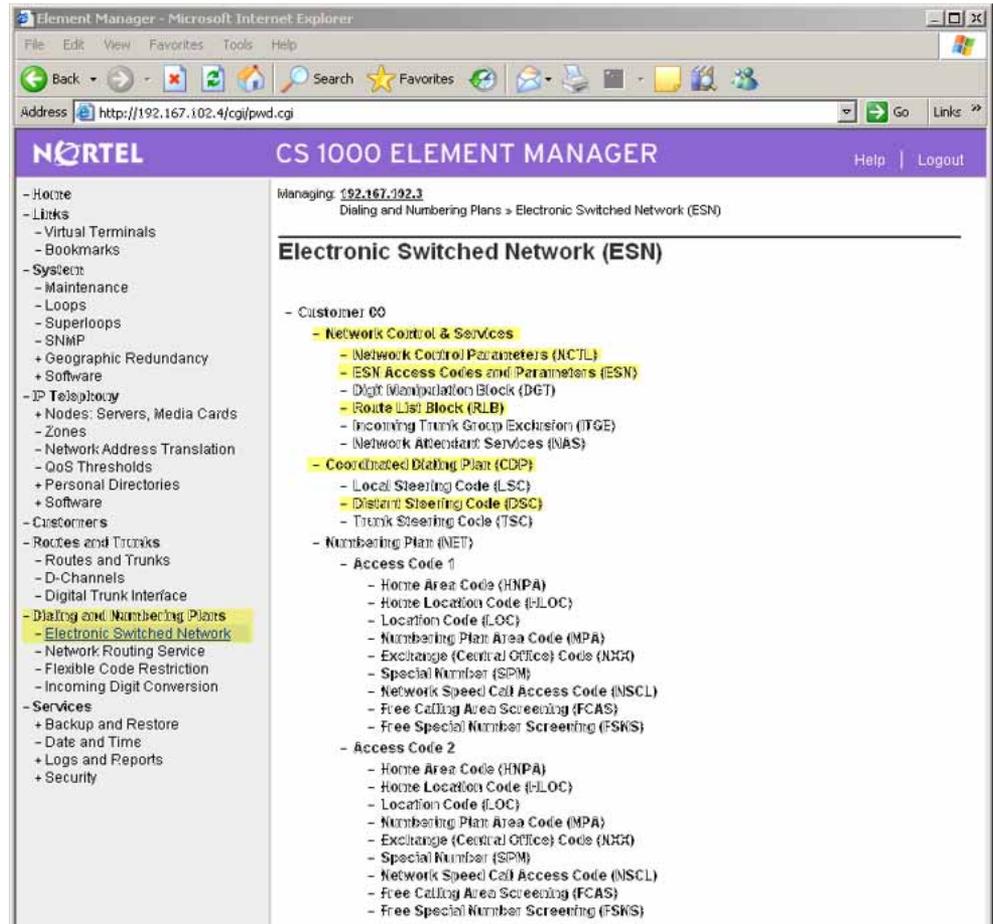
Creating the ESN data block for CDP

Perform the following procedure to create the ESN data block for CDP.

Creating the ESN data block for CDP

Step	Action
1	Log on to Element Manager.
2	Select Dialing and Numbering Plans > Electronic Switched Network .
3	Expand the Customer heading. See Figure 29 "Electronic Switched Network" (page 73).

Figure 29
Electronic Switched Network



- 4 Select **Network Control & Services > ESN Access Codes and Parameters (ESN)**.
- 5 A message appears if ESN data is not configured. Click **OK**. The ESN Access Codes and Basic Parameters page appears. See [Figure 30 "ESN Access Codes and Basic Parameters" \(page 74\)](#). If ESN data is configured on your switch, the fields on this page appear populated.

Figure 30
ESN Access Codes and Basic Parameters
ESN Access Codes and Basic Parameters

Input Description	Input Value
Maximum number of Digit Manipulation tables (MXDM):	100
Maximum number of Route Lists (MXRL):	100
Time of Day Schedules (TODS): (Items separated by a space)	0 00 00 23 59
Routing Controls (RTCL):	<input type="checkbox"/>
Click for Trunk Group Access Restrictions (TGAR):	<input type="checkbox"/>
NCOS Map (MXMAP): (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
Maximum number of Incoming Trunk Group exclusion tables (MXIX):	100
Maximum number of Free Calling area screening tables (MXFC):	100
Maximum number of Free Special number screening tables (MXFS):	100
One or two digit NARS/BARS Access Code 1 (AC1):	9
NARS/BARS Dial Tone after dialing AC1 or AC2 access codes (DLTN):	<input checked="" type="checkbox"/>
Expensive Route Warning Tone (ERWT):	<input checked="" type="checkbox"/>
Expensive Route Delay Time (ERDT):	6
Extended Time of Day schedule (ETOD):	
Maximum number of LOC codes (NARS only) (MXLC):	100
Maximum number of Special Common Carrier entities (MXSCC):	
One or two digit NARS Access Code 2 (AC2):	6
Coordinated Dialing Plan feature for this customer (CDP):	<input checked="" type="checkbox"/>
Maximum number of Steering Codes (MXSC):	120
Number of digits in CDP DN (DSC+DN or LSC+DN) (NCDP):	6

Submit Refresh Cancel

- 6 Edit the main parameters (**MXDM**, **MXRL**, **MXSD**, **MXIX**, **MXFC**, **MXFS** and **MXLC**) if required, or leave the default values as is.
- 7 Select the **Coordinated dialing Plan feature for this customer (CDP)** check box.
- 8 Set the value of the **Maximum number of Steering Codes (MXSC)**.
- 9 Set the value of the **Number of digits in CDP DN (DSC+DN or LSC+DN) (NCDP)**.

- 10 Click **Submit**.

—End—

Creating the Network Control Block (NCTL) for network access (LD 87)

Perform the following procedure to create the Network Control Block.

Creating the Network Control Block (NCTL) for network access (LD 87)

Step	Action
1	Log onto Element Manager.
2	Select Open Dialing and Numbering Plans > Electronic Switched Networks .
3	Expand the Customer tab. See Figure 29 "Electronic Switched Network" (page 73) .
4	Select Network Control and Services > Network Control Parameter (NTCL) . A message appears if no network control data is configured. Click OK to configure new data.
5	Next to Network Control Basic Parameters, click the Edit tab. The Network Control Basic Parameters page appears. See Figure 31 "Network Control Basic Parameters" (page 76) .

Figure 31
Network Control Basic Parameters

Element Manager - Microsoft Internet Explorer
Address: http://192.167.104.4/cgi/pwd.cgi

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 192.167.104.3
Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Network Control & Services > Network Control Parameters > Network Class of Service Group

Network Class of Service Group

Input Description	Input Value
Network Class of Service group number (NCOS):	0
Maximum Precedence Level (MPL):	
Equal Access associated with this NCOS group (EAG):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0
Expansive Route Warning Tone (RWTA):	<input type="checkbox"/>
Network Speed Call access allowed (NSC):	<input type="checkbox"/>
Off-Hook Queuing eligibility (OHQ):	<input type="checkbox"/>
Call Back Queuing eligibility (CBQ):	<input type="checkbox"/>
Starting Priority in CBQ (SPRL):	0
Maximum Priority attainable in CBQ (MPRL):	0
Priority Promotion timer (PROW):	0
MLPP service domain class of service (MLSD):	000000
ARDL network route selection (ARDL):	Allowed from ALL (both initial and extended) route sets (A)

Submit Refresh Cancel

6 If required, type a **Network Class of Service group number (NCOS)**.

7 Click **Submit**.

—End—

Creating the RLB for the virtual trunk route (LD 86)

Perform the following procedure to create the RLB for the virtual trunk route.

Creating the RLB for the virtual trunk route (LD 86)

Step	Action
------	--------

1	Log on to Element Manager.
---	----------------------------

- 2 Select **Dialing and Numbering Plans > Electronic Switched Networks**.
- 3 Expand the **Customer** heading.
See [Figure 29 "Electronic Switched Network"](#) (page 73).
- 4 Select **Network Control and Services > Route List Blocks (RLB)**.
If route list blocks are not configured, the error message "Route List does not exist" appears. Click **OK**.
- 5 Type the **Route List Index** number.
- 6 Click to **Add**.
The Route List Block Configuration page appears. See [Figure 32 "Route List Block"](#) (page 77).

Figure 32
Route List Block

Input Description	Input Value
Route List Index (RLI):	2
Entry Number for the Route List (ENR):	0
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	1
Skips Conventional Signaling (SCNV):	<input type="checkbox"/>
Use Tone Detector (TDET):	<input type="checkbox"/>
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VNS):	<input type="checkbox"/>
Conversion to LDN (CNV):	<input type="checkbox"/>
Expensive Route (EXP):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0
Digit Manipulation Index (DMI):	0
ISL D-Channel Down Digit Manipulation Index (ISDM):	0
Free Calling Area Screening Index (FCI):	0
Free Special Number Screening Index (FSNI):	0
Business Network Extension Route (BNE):	<input type="checkbox"/>
Strategy on Congestion (SBOC):	Reroute All (RRA)
- 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):	<input type="checkbox"/>
Off-Hook Queuing Allowed (OHQ):	<input type="checkbox"/>
Call Back Queuing Allowed (CBQ):	<input type="checkbox"/>
Number of Alternate Routing Attempts (NALT):	5
Initial Set (ISET):	0
Set Minimum Facility Restriction Level (MFRL):	
Overlap Length (OVL):	0

Submit Cancel

- 7 Select the **Route Number (ROUT)** you previously defined.
- 8 For **Strategy on Congestion (SBOC)**, select **Reroute All (RRA)**.
- 9 Accept the other defaults and click **Submit**.
The new Route List Block is generated. You can check the configuration by selecting Route List Block Index and Data Entry Index.

—End—

Creating the CDP steering codes (LD 87)

Perform the following procedure to create the CDP steering codes.

Creating the CDP steering codes (LD 87)

Step	Action
1	Log on to Element Manager.
2	Select Dialing and Numbering Plans > Electronic Switched Network .
3	Expand the Customer heading. See Figure 29 "Electronic Switched Network" (page 73) .
4	Select Coordinated Dialing Plan (CDP) > Distant Steering Code List .
5	Enter the Distant Steering Code (DSC) . This is the DN range of other systems on the network. You can add more steering codes in this manner.
6	Click to Add . The Distant Steering Code page appears. See Figure 33 "Distant Steering Code" (page 79) .

Figure 33
Distant Steering Code

Element Manager - Microsoft Internet Explorer
Address: http://192.167.102.4/cgi/pwd.cgi

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

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

Distant Steering Code

Input Description	Input Value
Distant Steering Code (DSC):	45
Flexible Length number of digits (FLEN):	0
Display (DSP):	Local Steering Code (LSC)
Remote Radio Paging Access (RRPA):	<input type="checkbox"/>
Route List to be accessed for trunk steering code (RLI):	1
Collect Call Blocking (CCBA):	<input type="checkbox"/>
maximum 7 digit NPA code allowed (NPA):	
maximum 7 digit NXX code allowed (NXX):	

Submit Refresh Delete Cancel

- 7 Check the populated fields.
- 8 Select a **Route list to be accessed for trunk steering code (RLI)**.
- 9 Click **Submit**.
- 10 Repeat steps 6 to 9 for all other call types on your network:
 - LOC (Location Code)
 - HLOC (Home Location Code)
 - NPA
 - HNPA (Home NPA)
 - SPN (Special Numbers)
 - NXX
- 11 This steering code is now defined. You can click the plus sign to view all the entered information.

—End—

Checking CODEC and QoS settings

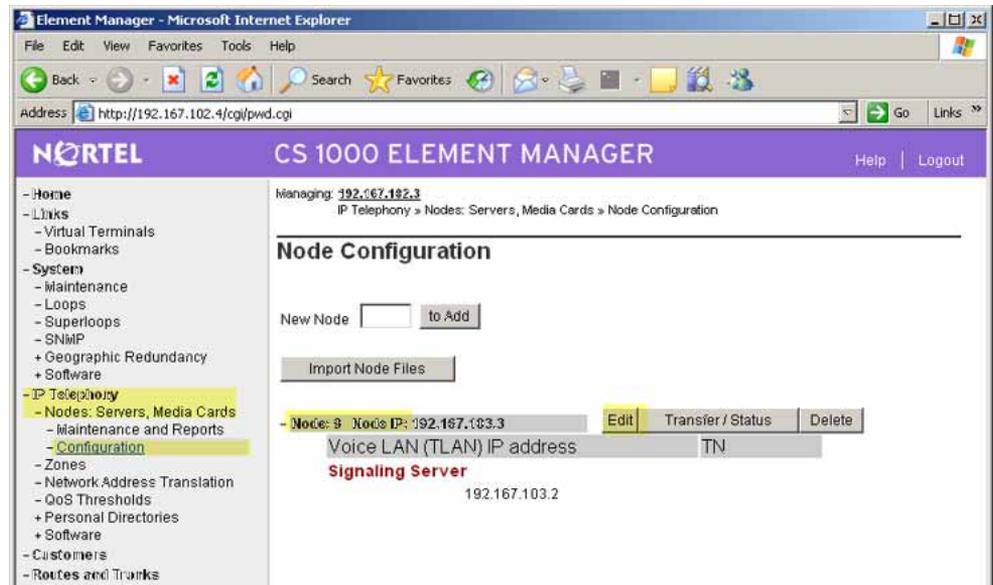
At this point, the Call Server configuration is complete. It is recommended that you check the CODEC and QoS settings.

Checking CODEC and QoS settings

Step	Action
------	--------

- | | |
|---|---|
| 1 | Log on to Element Manager. |
| 2 | Select IP Telephony Manager > Nodes: Servers, Media Cards > Configuration .
See Figure 34 "Node Configuration" (page 80). |

Figure 34
Node Configuration



- | | |
|---|--|
| 3 | Expand the Node heading. |
| 4 | Click Edit .
The Edit page appears. See Figure 35 " (page 81). |

Figure 35
Node Editing

Edit

Save and Transfer Cancel

- Mode

Mode ID 9

Voice LAN (TLAN) Node IP address 192.167.103.3

Management LAN (ELAN) gateway IP address 192.167.102.1

Management LAN (ELAN) subnet mask 255.255.255.0

Voice LAN (TLAN) subnet mask 255.255.255.0

+ SNMP Add

- VGW and IP phone codec profile

Enable Echo canceler

Echo canceler 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 payload 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

- 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

+ LAN configuration

+ SNMP

+ H323 GW Settings

+ Firmware

+ SIP GW Settings

+ SIP URD Map

+ SIP CD Services

+ SIP CTI Services

+ Cards Add

+ Signaling Servers Add

Save and Transfer Cancel

- 5 Expand the **VGW and IP phone CODEC profile** heading and edit the fields as necessary.
- 6 Expand the **QoS** heading and edit the fields as necessary.

- 7 If you make configuration changes, click **Save and Transfer**; otherwise, click **Cancel**.

—End—

NRS configuration

The Network Routing Service (NRS) uses a basic SIP structure for its configuration, which is applicable for SIP, H.323, and Network Connection Server (NCS) call completion. This structure is the basis of the single network dialing/numbering plan.

Before you begin configuration of the NRS, gather the names of all domains and subdomains.

NRS configuration procedures

The sequence of NRS configuration procedures is as follows:

- Launching NRS Manager
- Verifying and adjusting system-wide settings
- Configuring the NRS server settings (H.323 Gatekeeper or SIP)
- Configuring the service domain
- Configuring the L1 domain (UDP)
- Configuring the L0 domain (CDP)
- Configuring Gateway endpoints
- Configuring routing entries
- Configuring collaborative servers
- Updating the database
- Checking the status of registered endpoints
- Checking the status of virtual D-channels
- Checking the status of virtual trunks

Launching NRS Manager

Perform the following procedure to launch NRS Manager.

Launching NRS Manager

Step	Action
1	Log on to Element Manager.
2	Select Dialing and Numbering Plans .
3	Select Network Routing Service .
4	Click Next . The NRS logon page appears.
5	Enter the user ID and password. The NRS Overview page appears. See NRS Overview.

Figure 36
NRS Overview

Location: Home > NRS Overview >

> NRS Overview
System Wide Settings
NRS Server Settings

Network Routing Service		
Software version	sse-4.50.88	
Connected NRS role	PrimaryNRS	
Primary NRS IP (TLAN)	192.167.105.2	
Primary NRS state	ACTIVE	
Alternate NRS IP (TLAN)	Unknown	
Alternate NRS state	Unknown	
Alternate permanent in service	OFF	

Configured Components		
# of Service Domains	1	
# of L1 Domains (UDP)	1	
# of L0 Domains (CDP)	1	
# of Gateway Endpoints	2	
# of User Endpoints	0	
# of Routing Entries	4	
# of Default Routes	0	
# of Collaborative Servers	0	

Users Logged Into This NRS Manager		
admin	207.179.167.96	

—End—

Verifying and adjusting system-wide settings

You can check system-wide settings and make changes from NRS Manager.

Verifying and adjusting system-wide settings

- | Step | Action |
|------|--|
| 1 | Log on to NRS Manager. |
| 2 | Select System Wide Settings .
The System Wide Settings page appears. See System Wide Settings. |

Figure 37
System Wide Settings

Location: Home > System Wide Settings >

System Wide Settings

DB sync interval for alternate [Hours]

SIP registration time to live timer [Seconds]

H.323 gatekeeper registration time to live timer [Seconds]

H.323 alias name *

Alternate NRS server is permanent

Auto backup time [HH:MM]

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

- 3 Configure the information in the System Wide Settings page. Refer to System Wide Settings fields.

Table 4
System Wide Settings fields

Field	Description
DB Synch interval for alternate [Hours]	24 is the default.
SIP registration Time-to-Live timer	30 seconds is recommended.

Field	Description
H.323 Gatekeeper registration Time-to-Live timer	30 seconds is recommended.
H.323 Alias Name	This is a mandatory field. The H.323 Alias Name must be alphanumeric and contain no spaces. The default value is the same as the H.323 ID and HostName value configured in the PRIMARY Signaling Server's config.ini file.
Alternate NRS server is permanent	Select this check box if the Alternate NRS Server is to remain in service after a switch-over, even if the Primary NRS recovers. Clear the check box if the Alternate NRS switches over functions to the Primary NRS Server after the Primary NRS Server recovers.
Auto backup time	Enter the time when the database backup automatically occurs.
Auto backup to FTP site enabled	Select this check box to enable automatic backup of the NRS database to an FTP site.
Auto backup FTP site IP address Auto backup FTP site path Auto backup FTP site username Auto backup FTP site password	Enter values for Autobackup FTP if you enabled automatic backup of the NRS database to an FTP site.

4 Click **Save**.

—End—

Configuring the NRS server settings (H.323 Gatekeeper or SIP)

Perform the following procedure to configure NRS server settings.

Configuring the NRS server settings (H.323 Gatekeeper or SIP)

Step	Action
1	Log on to NRS Manager.
2	Select NRS Server Settings . The NRS Overview page appears. See NRS Overview.

Figure 38
NRS Overview

Location: Home > NRS Overview >

Network Routing Service	
Software version	sse-4.50.88
Connected NRS role	PrimaryNRS
Primary NRS IP (TLAN)	192.167.105.2
Primary NRS state	ACTIVE
Alternate NRS IP (TLAN)	Unknown
Alternate NRS state	Unknown
Alternate permanent in service	OFF
Configured Components	
# of Service Domains	1
# of L1 Domains (UDP)	1
# of L0 Domains (CDP)	1
# of Gateway Endpoints	2
# of User Endpoints	0
# of Routing Entries	4
# of Default Routes	0
# of Collaborative Servers	0
Users Logged Into This NRS Manager	
admin	207.179.167.96

- 3 Under **NRS Settings**, set the following values:
 - **Host name**
 - **Primary IP (T-LAN)**
 - **Alternate IP (T-LAN)**
 - **Control priority**
- 4 Under **H.323 Gatekeeper Settings**, select the **Location request (LRQ) response timeout**.
- 5 Under **SIP Server Settings**, set the following values:
 - **Mode**
 - **UDP transport enabled/disabled**
 - **UDP port**
 - **UDP maximum transmission unit (MTU)**
 - **TCP transport enabled/disabled**

- **TCP port**
- **TCP maximum transmission unit (MTU)**

Make the values under SIP Server Settings the same as those you configure for the SIP Proxy in Element Manager.

- 6 Under **Network Connection Server (NCS) Settings**, set the following values:
 - **Primary NCS port number**
 - **Alternate NCS port number**
 - **Primary NCS timeout**
- 7 Click **Save**.

—End—

Configuring the service domain

The NRS database information configured in this procedure is required by both the SIP Redirect Server and the H.323 Gatekeeper.

Configuring the service domain

Step	Action
1	Log on to NRS Manager.
2	Select the Configuration tab.
3	Click Standby DB View to switch from active to standby database view. The active database view is the default view. Use the active database for runtime queries, and the standby database for administrator modifications. You must use standby view to make changes to the database. See Service Domains.

Figure 39
Service Domains

#	ID	Description	# of L1 domains	# of L0 domains	# of gateway endpoints
1	cdsig.com	Not available	1	1	2

- 4 Select **Service Domains**.
- 5 Click **Add**.
- 6 Enter your **Domain name** and a **Domain description**.
These values must match that set for the Signaling Server.
- 7 Click **Save**.
The Service Domains page appears again with the new domain added.
When no description is entered, the service domain is shown with the message “Not available”. This means that the description is not entered, but the service domain is still active. This applies to all description fields in NRS Manager.

—End—

Configuring the L1 domain (UDP)

You can configure the L1 domain after you configure the service domain. The L1 domain is a service domain associated with UDP.

Configuring the L1 domain (UDP)

Step	Action
1	Log on to NRS Manager.
2	Select the Configuration tab.
3	Click Standby DB View to switch from active to standby database view.

- 4 Select **L1 Domains**.
- 5 Click **Add**.
The View L1 Domain Property page appears. See View L1 Domain Property .

Figure 40
View L1 Domain Property

Location: Configuration > L1 Domains (UDP) > View L1 Domain Property >

View L1 Domain Property (cdsig.com)

Domain name	<input type="text" value="udp"/>	*
Domain description	<input type="text"/>	
Endpoint authentication enabled	<input type="text" value="Authentication off"/>	
Authentication password	<input type="text"/>	
E.164 country code	<input type="text" value="1"/>	*
E.164 area code	<input type="text" value="506"/>	*
E.164 international dialing access code	<input type="text" value="011"/>	
E.164 national dialing access code	<input type="text" value="9"/>	
E.164 local (subscriber) dialing access code	<input type="text" value="6"/>	
Private L1 domain (UDP location) dialing access code	<input type="text" value="6"/>	
Special number	<input type="text" value="9"/>	
Emergency service access prefix	<input type="text" value="9"/>	
Special number label	<input type="text" value="PrivateSpecial"/>	

** Mandatory field indicator*

- 6 Configure the L1 domain.

Refer to L1 domain fields for configuration information.

Table 5
L1 domain fields

Field	Value	Description
Domain name	<alphanumeric string>	Mandatory. The name must be alphanumeric and can be up to 30 characters in length.
Domain description	<character string>	Optional. The description can include any character except single quotes and be up to 120 characters in length.
Endpoint authentication enabled	Authentication off Authentication on	If Authentication on is selected, all endpoints require authentication.
Authentication password	<alphanumeric string>	If Authentication on is selected, enter an authentication password. The password must be alphanumeric and up to 30 characters in length.
E.164 country code	<numeric string>	Mandatory. The code must be numeric and up to 7 characters in length.
E.164 area code	<numeric string>	Mandatory. The code must be numeric and up to 7 characters in length.
E.164 International Dialing Access Code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 national dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 local (subscriber) dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Private L1 domain (UDP location) dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Special number	<numeric string>	Optional. The number must be numeric and up to 30 characters in length.
Emergency service access prefix	<numeric string>	Optional. The number must be numeric and up to 30 characters in length.
Special number label	<alphanumeric string>	Optional. The label must be alphanumeric and up to 30 characters in length. The first character in the label must be alphabetic.

- 7 Click **Save**.
The L1 Domains page appears again with the new L1 domain added.

- 8 To view the configured data for the L1 domain, click the **ID** in the **L1 Domains (UDP)** menu.
The View L1 Domain Property page appears, displaying your configured data.

—End—

Configuring the L0 domain (CDP)

The L0 domain is a service domain associated with CDP, representing the private addresses field in the SIP URI. This address is composed of the service, L1, and L0 domains.

Configuring the L0 domain (CDP)

Step	Action
1	Log on to NRS Manager.
2	Select the Configuration tab.
3	Click Standby DB View to switch from active to standby database view.
4	Select L0 Domains .
5	Click Add . The View L0 Domain Property page appears. See View L0 Domain Property.

Figure 41
View L0 Domain Property

Location: Configuration > L0 Domains (CDP) > View L0 Domain Property >

View L0 Domain Property (cdsig.com / udp)

Domain name	<input type="text" value="cdp"/>	*
Domain description	<input type="text"/>	
Endpoint authentication enabled	<input type="text" value="Not configured"/>	
Authentication password	<input type="text"/>	
E.164 country code	<input type="text" value="1"/>	
E.164 area code	<input type="text" value="506"/>	
Private unqualified number label	<input type="text" value="PrivateUnknown"/>	
E.164 international dialing access code	<input type="text" value="011"/>	
E.164 national dialing access code	<input type="text" value="9"/>	
E.164 local (subscriber) dialing access code	<input type="text" value="6"/>	
Private L1 domain (UDP location) dialing access code	<input type="text" value="6"/>	
Special number	<input type="text" value="9"/>	
Emergency service access prefix	<input type="text" value="9"/>	

* Mandatory field indicator

- 6 Enter the appropriate values for your network.
Refer to Add L0 Domain fields for configuration information.

The country codes, area codes, public prefixes, and private prefixes must match those of your L1 domain.

Table 6
Add L0 Domain fields

Field	Value	Description
Domain name	<alphanumeric string>	Mandatory. The name must be alphanumeric and can be up to 30 characters in length.
Domain description	<character string>	Optional. The description can include any character except single quotes and can be up to 120 characters in length.
Endpoint authentication enabled	Authentication off Authentication on	If Authentication on is selected, then all endpoints require authentication.
Authentication password	<alphanumeric string>	if Authentication on is selected, enter a password. The password must be alphanumeric and up to 30 characters in length.
E.164 country code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 area code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Private unqualified number label	<alphanumeric string>	The label must be alphanumeric and up to 30 characters in length. The first character in the label must be alphabetic.
E.164 international dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 national dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 local (subscriber) dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Private L1 domain (UDP) location) dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Special number	<numeric string>	Optional. The number must be numeric and up to 30 characters in length.
Emergency service access prefix	<numeric string>	Optional. The number must be numeric and up to 30 characters in length.

- 7 Click **Save**.
The L0 Domains page appears again, showing the added domain.
- 8 To view the configured data, in the L0 domain page, select the service domain and L1 domain and click **Show**.
A list of configured L0 domains appears.
- 9 Select the L0 domain to view its configuration information.
The View L0 Domain Property page appears. See View L0 Domain Property.

—End—

Configuring Gateway endpoints

Add an endpoint for both the Communication Server 1000 and Multimedia Communication Server 5100 (MCS 5100).

These are Gateway endpoints, which can be served by several users. You can add multiple endpoints, some SIP-only, some H.323-only, and Unistim. You can also configure a user endpoint, which can be only one of these three protocols.

Configuring Gateway endpoints

Step	Action
1	Log on to NRS Manager.
2	Select the Configuration tab
3	Click Standby DB View to switch from active to standby database view.
4	Click Gateway Endpoints .
5	Click Add . The View Gateway Endpoint Property page appears. See View Gateway Endpoint Property.

Figure 42
View Gateway Endpoint Property

Location: Configuration > Gateway Endpoints > View Gateway Endpoint Property >

View Gateway Endpoint Property (cdsig.com / udp / cdp)

Endpoint name *

Endpoint description

Tandem gateway endpoint name [Look up](#)

Endpoint authentication enabled

Authentication password

E.164 country code

E.164 area code

E.164 international dialing access code

E.164 national dialing access code

E.164 local (subscriber) dialing access code

Private L1 domain (UDP location) dialing access code

Private special number 1

Private special number 2

Static endpoint address type

Static endpoint address

H.323 Support

SIP support

SIP transport

SIP port

Network Connection Server enabled

* Mandatory field indicator

- 6 Enter the appropriate values for your network.
Refer to Add Gateway Endpoint fields for configuration information.

Table 7
Add Gateway Endpoint fields

Field	Value	Description
Endpoint name	<alphanumeric string>	The name must be alphanumeric and up to 30 characters in length. Note: Configure the MCS 5100 Gateway endpoint name as convergeddesktop .
Endpoint Description	<alphanumeric string>	The description must be alphanumeric and up to 120 characters in length.

Field	Value	Description
Tandem Gateway endpoint name	<alphanumeric string>	The tandem Gateway is optional. This indicates whether the endpoint is used to tandem calls from outside the network. The name must be alphanumeric and up to 30 characters in length. Note: Use the Look-up link to find configured Gateway endpoints.
Endpoint authentication enabled	Not configured Authentication off Authentication on	If this option is selected, the Gateway endpoint uses the L1 or L0 authentication (if enabled). If this option is selected, authentication is off for this Gateway endpoint even if L1 or L0 authentication is enabled. If this option is selected, authentication is on for this Gateway endpoint, and the authentication overrides the L1 or L0 authentication (if enabled).
Authentication password	<alphanumeric string>	If Authentication on is selected, choose a password. The password must be alphanumeric and up to 30 characters in length.
E.164 country code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 area code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 international dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 national dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
E.164 local (subscriber) dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Private L1 domain (UDP location) dialing access code	<numeric string>	Optional. The code must be numeric and up to 7 characters in length.
Private special number 1	<numeric string>	Optional. The number must be numeric and up to 30 characters in length.
Private special number 2	<numeric string>	Optional. The number must be numeric and up to 30 characters in length.
Static endpoint address type	IP version 4	Select IP version 4 from the drop-down list.
Static endpoint address	<Node IP address>	This is the address of the MCS 5100 application server, responsible for the MCS voice path. If a third-party Gateway is used, it is the IP address of the Gateway.

Field	Value	Description
H.323 support	H.323 not supported RAS H.323 endpoint Not RAS H.323 endpoint	RAS H.323 Endpoint is recommended. If an H.323 Gateway Endpoint is configured with an H.323 Support type of RAS H.323 endpoint, NRS Manager displays Endpoint Dynamic Registration information after the H.323 Gateway registers with the NRS. Endpoint Dynamic Registration information includes Call Signaling IP, RAS IP, Alias name, t35Country code, t35Extension, Manufacturer code, Product ID, and Version ID.
SIP support	SIP not supported Static SIP endpoint Dynamic SIP endpoint	Dynamic SIP Endpoint is recommended. If a SIP Trunk Gateway Endpoint is configured with a SIP Support type of Dynamic SIP endpoint, NRS Manager displays Endpoint Dynamic Registration Information for SIP after the SIP Trunk Gateway registers with the NRS. Endpoint Dynamic Registration Information includes SIP IP, Registration expiry time, User agent, and Preference.
SIP transport	TCP UDP	TCP is selected by default. This information should match the SIP Transport Protocol in the Signaling Server Properties.
SIP port	<port value>	Default SIP port value is 5060. If the SIP Port is changed, the value must be numeric and up to 5 numbers in length. The range is 0 to 65535.
Network Connection Server enabled	<check box>	Select the Network Connection Server is enabled check box if this Gateway Endpoint supports the NCS for branch office or SRG user redirection to the main office, Virtual Office, or Geographic Redundancy.

- 7 Click **Save**.
The Gateway Endpoints window appears again with the new endpoints added.
- 8 From the Gateway Endpoint page, select the **Service, Domain, L1 domain**, and **L0 domain** to view the configured data of an endpoint.
- 9 Click **Show**.
A list of configured Gateways appears.
- 10 Click the **Gateway ID** you wish to view.

The View Gateway Endpoint Property page appears. See View Gateway Endpoint Property.

—End—

Configuring routing entries

Perform the following procedure to configure routing entries.

Configuring routing entries

Step	Action
1	Log on to NRS Manager.
2	Select the Configuration tab
3	Click Standby DB View to switch from active to standby database view.
4	Click Routing Entries .
5	Type the relevant Gateway endpoint OR Click the Look up link and perform a search. All configured endpoints appear.
6	Click the endpoint to configure as a Routing Entry.
7	Select a DN type. For UDP , select Private level 1 regional (UDP location code) .
8	For CDP , select Private level 0 regional (CDP steering code) .
9	Click Show . The Routing Entries page appears. See Routing Entries.

Figure 43
Routing Entries

Location: Configuration > Routing Entries >

Routing Entries

Show Routing Entries for (Service Domain / L1 Domain / L0 Domain / Endpoint)
 Select domains and enter a gateway endpoint name to show specified routing entries.
 Use the wildcard * by itself for all gateway endpoints :

/
 /
 /

Gateway Endpoint: [Look up](#)

With DN Type:

Showing 1 - 3 of 3 < Previous Next >				
#	DN Prefix	DN Type	Route Cost	SIP URI Phone Context
1	<u>22</u>	Private level 0 regional (CDP steering code)	1	cdp.udp
2	<u>224350</u>	Private level 0 regional (CDP steering code)	1	cdp.udp
3	<u>23</u>	Private level 0 regional (CDP steering code)	1	cdp.udp

- 10 Click **Add** to add a new Routing Entry. The View Routing Entry Property page appears. See View Routing Entry Property .

Figure 44
View Routing Entry Property

Location: Configuration > Routing Entries > View Routing Entry Property >

** Mandatory field indicator*

- 11 Enter the **DN prefix**.
This is the CDP DSC steering code.
- 12 Enter the **Route cost**.
The higher the number, the higher the cost. This is equivalent to Least Cost Routing.
- 13 Click **Save**.

—End—

Configuring collaborative servers

A Collaborative Server is a server in another network zone that can resolve requests when your NRS cannot find a match in its numbering plan database.

You can specify a list of Networking Routing Servers in different network zones in your NRS. The NRS Manager provides a utility for adding and viewing a list of NRSs (Collaborative Servers) in different network zones.

Configuring collaborative servers

Step	Action
1	Log on to NRS Manager.
2	Select the Configuration tab
3	Click Standby DB View to switch from active to standby database view.

- 4 Click **Collaborative Servers**.
- 5 Click **Add**.
The Add Collaborative Server page appears. See Add Collaborative Server. This page may differ from the view shown here depending on the value you choose for the Domain type for collaborative server.

Figure 45
Add Collaborative Server

Location: Configuration > Collaborative Servers > Add Collaborative Server >

Add Collaborative Server

Domain type for collaborative Server L1 domain ▾

L1 domain name (with service domain path) cdsig.com / udp ▾

Alias name

Server address type IP version 4 ▾

Server address *

H.323 support

RAS port

SIP support

SIP transport TCP ▾

SIP port

Network Connection Server support

Network Connection Server transport UDP ▾

Network Connection Server port

* Mandatory field indicator

- 6 For **Domain type for collaborative Server**, select either **L0 domain** or **L1 domain**.
- 7 For the **L1** or **L0 domain name**, select the L1 or L0 domain name.
- 8 Enter the **Alias name** of the collaborative server.
The alias name must be alphanumeric and contain no spaces.

- 9 For **Server address type**, select **IP version 4**.
- 10 For **Server address**, type the server IP address.
- 11 If the H.323 protocol is supported by the server, populate the following fields:
 - **H.323 support** check box
 - **RAS port**
- 12 If the SIP protocol is supported by the server, populate the following fields:
 - **SIP support** check box
 - **SIP transport** protocol
 - **SIP port**
- 13 If Network Connection Service is supported by the server, populate the following fields:
 - **Network Connection Server support** check box
 - **Network Connection Server transport**
 - **Network Connection Server port**
- 14 Click **Save**.
The Collaborative Servers page appears with the new server.
- 15 For redundancy purposes, perform this procedure again for the alternate NRS in the other network zone.

—End—

Updating the database

To save your entries, you must update the database.

Updating the database

Step	Action
1	Log on to NRS Manager.
2	Click the Tools tab.
3	Click the Database Actions tab. The Database Actions page appears, showing the Database State as Changed. See Database Actions.

Figure 46
Database Actions

Location: Tools > Database Actions >

- 4 From the **Select database action** menu, select **Cut over & Commit**.
- 5 Click **Submit**.

—End—

Checking the status of registered endpoints

Perform the following procedure to check the status of registered endpoints.

Checking the status of registered endpoints

Step	Action
------	--------

- | | |
|---|-------------------------------------|
| 1 | Log on to NRS Manager. |
| 2 | Click the Configuration tab. |
| 3 | Select Service Domains . |

See Service Domains.

- 4 Ensure that **Active DB View** is selected.
- 5 Click the number in the **# of Gateway endpoints** column.
- 6 Click **Show**.
The Gateway Endpoints page appears. See Gateway Endpoints.

Figure 47
Gateway Endpoints

Location: Configuration > Gateway Endpoints >

Gateway Endpoints					
Show Gateway Endpoints for (Service Domain / L1 Domain / L0 Domain):					
<input type="text" value="cdsig.com"/> / <input type="text" value="udp"/> / <input type="text" value="cdp"/> <input type="button" value="Show"/>					
Showing 1 - 2 of 2 < Previous Next >					
#	ID	Support Protocol(s)	Call Signaling IP	Description	# of routing entries
1	CS1000S_CP	RAS H.323 / Dynamic SIP / NCS	192.167.105.3 / 192.167.105.3	Not available	1
2	convergeddesktop	Static SIP	192.168.248.13	Converged Desk...	3

—End—

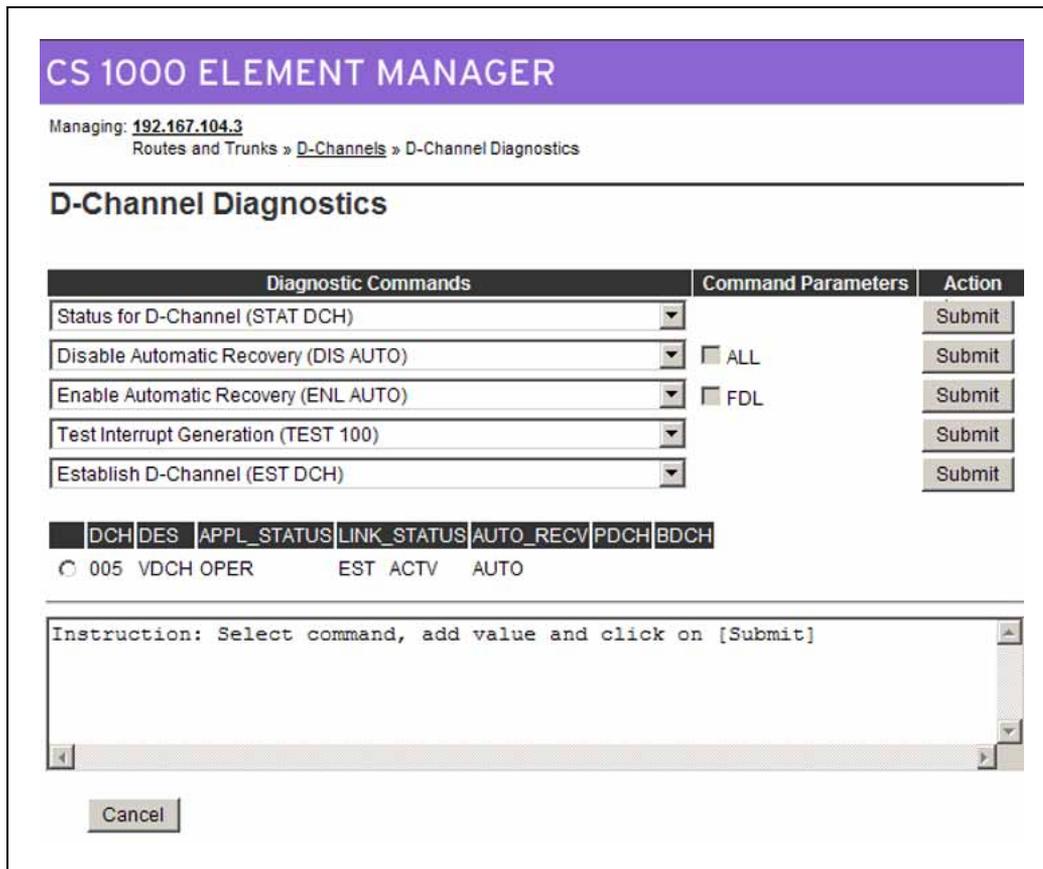
Checking the status of virtual D-channels

Perform the following procedure to check the status of virtual D-channels.

Checking the status of virtual D-channels

Step	Action
1	Log on to Element Manager.
2	Select Routes and Trunks > D-Channels .
3	Select D-Channel Diagnostics (LD 96) . The D-Channel Diagnostics page appears. See D-Channel Diagnostics.

Figure 48
D-Channel Diagnostics



- 4 Under **Diagnostic Commands**, select **Status for D-Channel (STAT DCH)**.
- 5 Click **Submit**.
- 6 Check that the D-Channel is operational, established, and active.

—End—

Checking the status of virtual trunks

Perform the following procedure to check the status of virtual trunks.

Checking the status of virtual trunks

Step	Action
------	--------

- | | |
|---|----------------------------|
| 1 | Log on to Element Manager. |
|---|----------------------------|

- 2 Select **IP Telephony > Nodes: Servers, Media Cards > Maintenance and Reports**.
- 3 Expand the **Node ID** heading.
- 4 Click **GEN CMD** for the switch.
The General Commands page appears. See General Commands.

Figure 49
General Commands

CS 1000 ELEMENT MANAGER

Managing: [192.167.104.3](#)
 IP Telephony » Nodes: Servers, Media Cards » [Node Maintenance and Reports](#) » General Commands

General Commands

Element IP: 192.167.104.4 Element Type: SS

Group	Vtrk	Command	vtrkShow	Protocol	SIP	Start	<input type="checkbox"/>	Range	<input type="checkbox"/>	RUN
IP address	192.167.104.3	Number of Pings	3	PING						

```

-----
VTRK Summary
-----
VTRK status   : Active
Protocol      : SIP
D-Channel     : 5
Customer      : 0
Channels Idle : 12
Channels Busy : 0
Channels Mbsy: 0
Channels Pend : 0
Channels Dsbl: 0
Channels Ukwn : 0
Channels Total: 12
Chid ranges   : 1 to 112
-----

```

IND	TN	DCH	PROTOCOL	CHID	CUST	ROUTE	MEMB	ICOG	VoIP
0	065-00	005	MCDN->EST	001	00	001	001	IO	SIP
1	065-01	005	MCDN->EST	002	00	001	002	IO	SIP
2	065-02	005	MCDN->EST	003	00	001	003	IO	SIP
3	065-03	005	MCDN->EST	004	00	001	004	IO	SIP
4	065-04	005	MCDN->EST	005	00	001	005	IO	SIP
5	065-05	005	MCDN->EST	006	00	001	006	IO	SIP
6	065-06	005	MCDN->EST	007	00	001	007	IO	SIP
7	065-07	005	MCDN->EST	008	00	001	008	IO	SIP
8	065-08	005	MCDN->EST	009	00	001	009	IO	SIP
9	065-09	005	MCDN->EST	010	00	001	010	IO	SIP
20	067-00	005	MCDN->EST	111	00	003	001	IO	SIP
21	067-01	005	MCDN->EST	112	00	003	002	IO	SIP

```

VTRK State = Active
-----
VTRK Status = Enabled
-----

```

- 5 For **Group**, select **Vtrk**.
- 6 For **Command**, select **vtrkShow**.

- 7 For **Protocol**, type **H323**.
- 8 Click **Run**.
The Virtual Trunk status appears.
- 9 Check each IP Phone manually. At the **DEF GW** option, verify that the IP address is the same as the T-LAN Gateway of that system. If the phones ring and have dial tone but there is no speech path, the Default Gateway is 0.0.0.0 and is not operational.

—End—

BCM 200/400 configuration

This chapter describes configuration procedures for the Business Communications Manager (BCM) 200 and 400 systems.

Element Manager as viewed on your system may differ slightly from the screens shown in this chapter because you can customize the column display in Element Manager.

BCM 200/400 configuration procedures

The sequence of BCM 200/400 configuration procedures is as follows:

- Configuring incoming VoIP trunks
- Verifying system license and keycodes
- Configuring VoIP trunk media parameters
- Configuring local Gateway parameters
- Configuring VoIP lines
- Configuring target lines

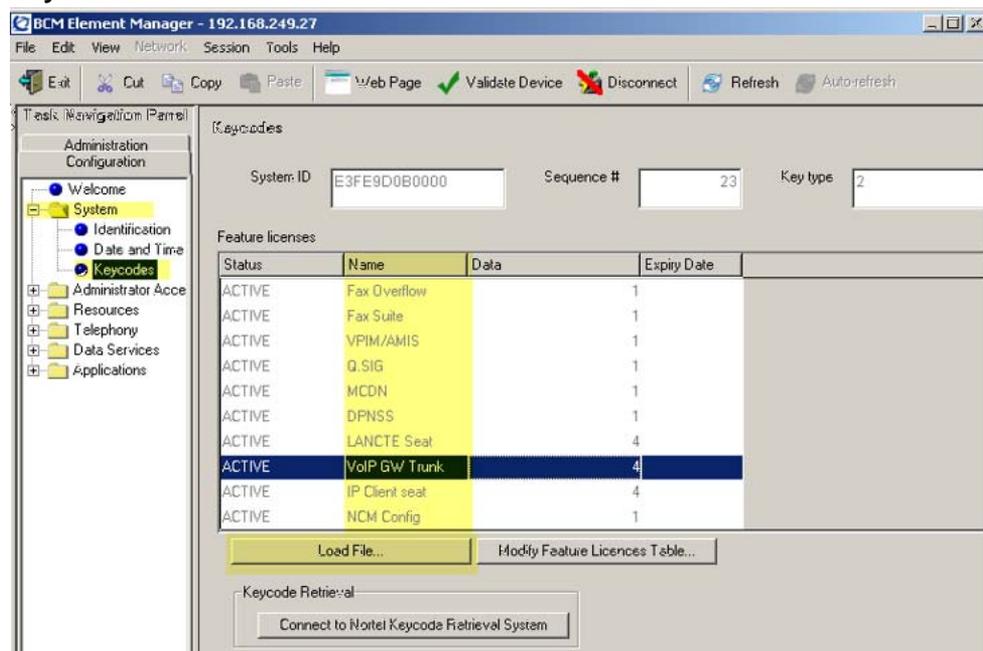
Configuring incoming VoIP trunks

Perform the following procedure to configure incoming VoIP trunks.

Configuring incoming VoIP trunks

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select System > Keycodes . See Keycodes.

Figure 50
Keycodes



- 4 Load new Keycodes by loading a new keycode file or connecting to Nortel's Keycode Retrieval System (KRS). For more information about keycodes and keycode retrieval, see *Keycode Installation Guide* (NN40010-301).

—End—

Verifying system license and keycodes

Perform the following procedure to verify system license and keycodes.

Verifying system license and keycodes

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select System > Keycodes . See Keycodes.
4	In the Name column, scroll down to VoIP GW Trunk . The number of license keys you have are listed in the Data column.

—End—

Configuring VoIP trunk media parameters

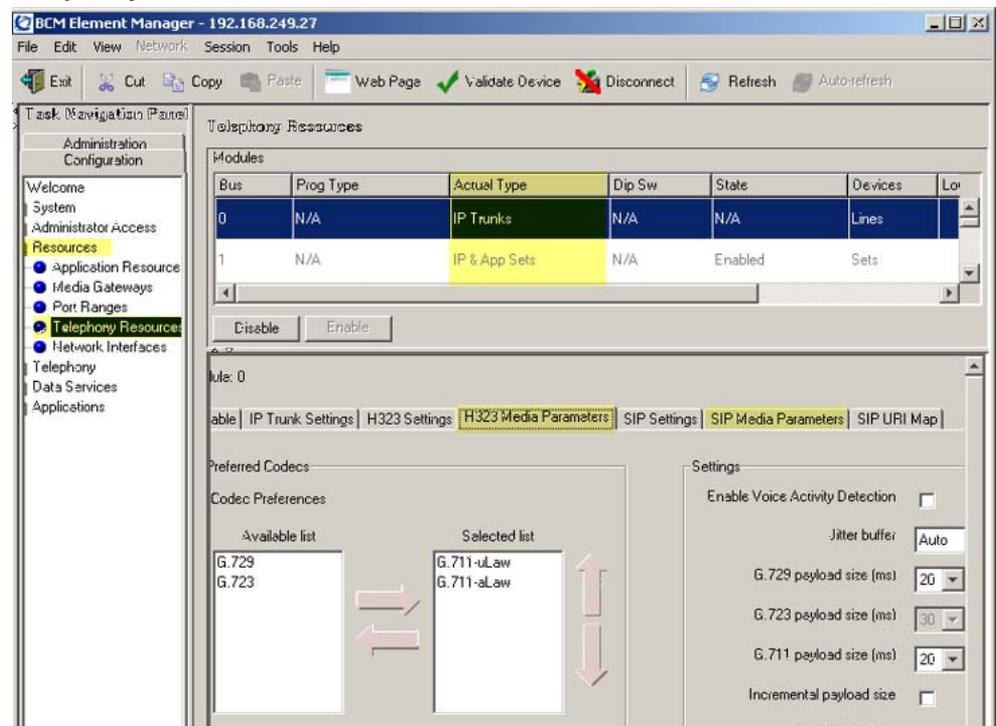
Perform the following procedure to configure VoIP trunk media parameters.

Configuring VoIP trunk media parameters

Step	Action
------	--------

- | | |
|---|--|
| 1 | Log on to Element Manager. |
| 2 | In the Task Navigation Panel , select the Configuration tab. |
| 3 | Select Resources > Telephony Resources .
See Telephony Resources. |

Figure 51
Telephony Resources



- | | |
|---|---|
| 4 | In the Modules panel, select the line where the Actual Type column is set to IP Trunks . |
| 5 | Select the H.323 Media Parameters or SIP Media Parameters tab. |
| 6 | Enter the information that supports your system. |

Ensure that these settings are consistent with the other systems on your network.

Refer to H.323 Media Parameters fields and SIP Media Parameters fields for a description of the parameters.

—End—

Table 8
H.323 Media Parameters fields

Field	Value	Description
Preferred Codecs	G.711 -uLaw G.711 -aLaw G.729 G.723	<p>Add codecs to the Selected list and order them in the order in which you want the system to attempt to use them. The system attempts to use the codecs in top-to-bottom sequence.</p> <p>Performance note: Codecs on all networked BCMs must be consistent to ensure the proper functionality of interacting features such as Transfer and Conference.</p> <p>Systems running BCM Release 3.5 or later allow codec negotiation and renegotiation to accommodate inconsistencies in codec settings over VoIP trunks.</p>
Enable Voice Activity Detection	<check box>	<p>Voice Activity Detection (VAD), also known as silence suppression, identifies periods of silence in a conversation and stops sending IP speech packets during those periods. In a typical telephone conversation, most of the conversation is half-duplex, meaning that one person is speaking while the other is listening. If VAD is enabled, no voice packets are sent from the listener end. This greatly reduces bandwidth requirements. G.723.1 and G.729 support VAD. G.711 does not support VAD.</p> <p>Performance note: VAD on all networked BCMs and IPT systems must be consistent to ensure functionality of features such as Transfer and Conference. The Payload size on the IPT must be set to 30ms.</p>
Jitter Buffer	Auto None Small Medium Large	Select the size of jitter buffer for your system.

Field	Value	Description
G.729 payload size (ms)	10,20,30,40,50,60	Set the maximum required payload size, per codec, for the VoIP calls sent over H.323 trunks. Note: Payload size can also be set for Nortel IP telephones. See <i>BCM 4.0 Telephony Device Installation Guide</i> (N0060609).
G.723 payload size (ms)	30	
G.711 payload size (ms)	10,20,30,40,50,60	
Incremental payload size	<check box>	When enabled, the system advertises a variable payload size (40, 30, 20, 10 ms).
Enable T.38 fax	<check box>	When enabled, the system supports T.38 fax over IP. Caution: Fax tones broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of the fax machine. To minimize the possibility of your VoIP calls being dropped due to fax tone interference: <ul style="list-style-type: none"> • place the fax machine away from other telephones • turn the fax machine's speaker volume to the lowest level, or off, if available
Force G.711 for 3.1k Audio	<check box>	When enabled, the system forces the VoIP trunk to use the G.711 codec for 3.1k audio signals, such as modem or TTY machines. Note: You also can use this setting for fax machines if T.38 fax is not enabled on the trunk.

Table 9
SIP Media Parameters fields

Field	Value	Description
Preferred Codecs	G.711 -uLaw G.711 -aLaw G.729 G.723	Add codecs to the Selected list and order them in the order in which you want the system to attempt to use them. The system attempts to use the codecs in a top-to-bottom sequence. Performance note: Codecs on all networked BCMs must be consistent to ensure the proper functionality of interacting features such as Transfer and Conference. Systems running BCM Release 3.5 or later allow codec negotiation and renegotiation to accommodate inconsistencies in codec settings over VoIP trunks.

Field	Value	Description
Enable Voice Activity Detection	<check box>	<p>Voice Activity Detection (VAD), also known as silence suppression, identifies periods of silence in a conversation and stops sending IP speech packets during those periods. In a typical telephone conversation, most of the conversation is half-duplex, meaning that one person is speaking while the other is listening. If VAD is enabled, no voice packets are sent from the listener end. This greatly reduces bandwidth requirements. G.723.1 and G.729 support VAD. G.711 does not support VAD.</p> <p>Performance note: VAD on all networked BCMs and IPT systems must be consistent to ensure functionality of features such as Transfer and Conference. The Payload size on the IPT must be set to 30ms.</p>
Jitter Buffer	Auto None Small Medium Large	Select the size of jitter buffer for your system.
G.729 payload size (ms) G.723 payload size (ms) G.711 payload size (ms)	10,20,30,40,50,60 30 10,20,30,40,50,60	<p>Set the maximum required payload size, per codec, for the VoIP calls sent over H.323 trunks.</p> <p>Note: Payload size can also be set for Nortel IP telephones. See <i>BCM 4.0 Telephony Device Installation Guide</i> (N0060609).</p>
Enable T.38 fax	<check box>	<p>When enabled, the system supports T.38 fax over IP.</p> <p>Caution: Fax tones broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of the fax machine. To minimize the possibility of your VoIP calls being dropped due to fax tone interference:</p> <ul style="list-style-type: none"> place the fax machine away from other telephones turn the fax machine's speaker volume to the lowest level, or off, if available

Configuring local Gateway parameters

Perform the following procedure to configure local Gateway parameters.

Configuring local Gateway parameters

- | Step | Action |
|------|--|
| 1 | Log on to Element Manager. |
| 2 | In the Task Navigation Panel , select the Configuration tab. |
| 3 | In the Module Panel , select the line in which the Actual Type column is set to IP Trunks .
See Telephony Resources. |
| 4 | Select the IP Trunk Settings tab and enter the information that supports your system.
See IP Trunk Settings. Refer to IP Trunk Settings fields for information about the IP Trunk Settings fields. |

Figure 52
IP Trunk Settings

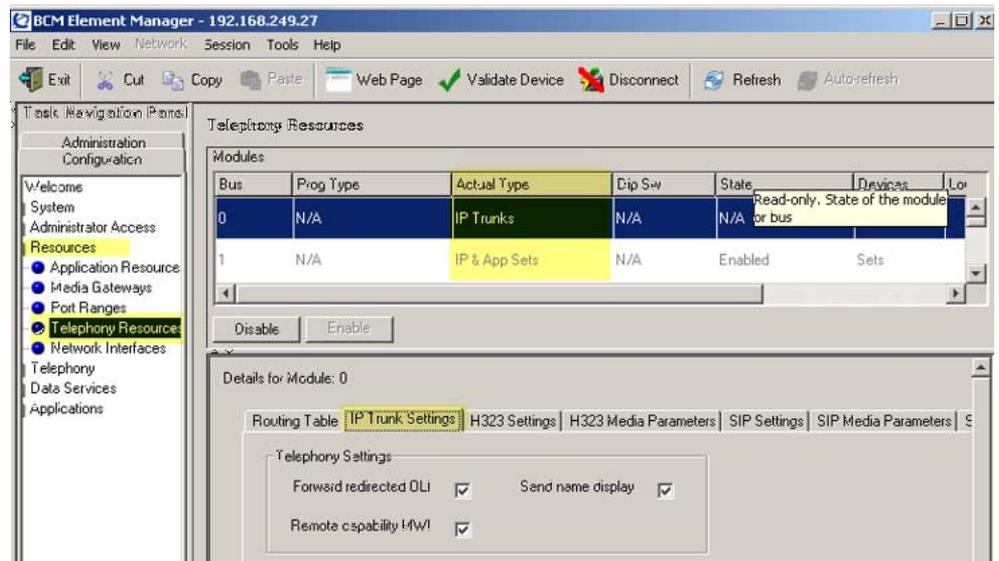


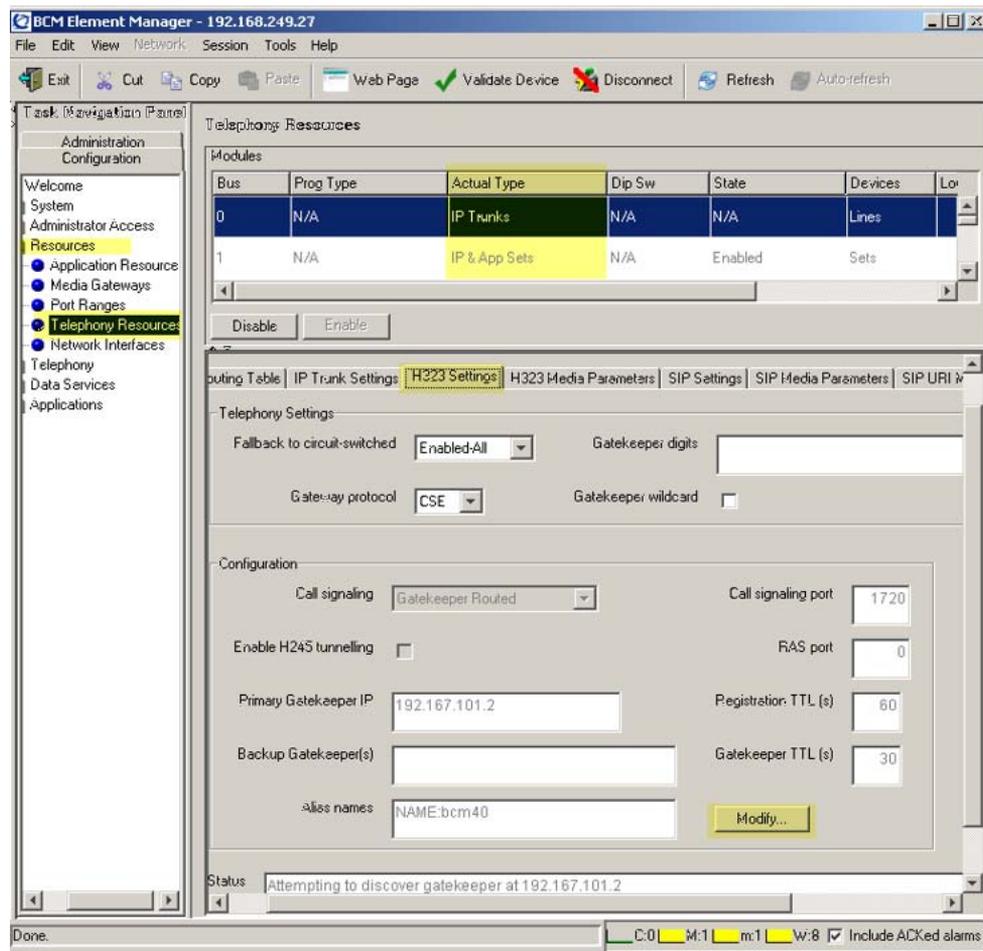
Table 10
IP Trunk Settings fields

Field	Value	Description
Forward redirected OLI	<check box>	If enabled, the OLI of an internal telephone is forwarded over the VoIP trunk when a call is transferred to an external number over the private VoIP network. If disabled, only the CLID of the transferred call is forwarded.

Field	Value	Description
Send name display	<check box>	If enabled, the telephone name is sent with outgoing calls to the network.
Remote capability MWI	<check box>	This setting must coordinate with the functionality of the remote system hosting remote voice mail.

- 5 For H.323 VoIP trunks, select the **H.323 Settings** tab. See H.323 Settings.

Figure 53
H.323 Settings



- 6 When implementing your dialing plan, in the **H.323 Settings** tab, select a value for **Fall back to circuit-switched**. This determines how the system handles calls if the IP network cannot be used.
- 7 For **Gateway protocol**, select **CSE**.
- 8 Scroll down to **Alias Name** and click **Modify**.

The Modify Call Signaling Settings page appears.

- 9 Enter the information that supports your system.
Applying the changes made to the Call Signaling Settings causes all H.323 calls to be dropped. It is recommended that you make changes to the Call Signaling Settings during off-peak hours or a scheduled maintenance window.
Refer to H.323 Call Signaling Settings fields.

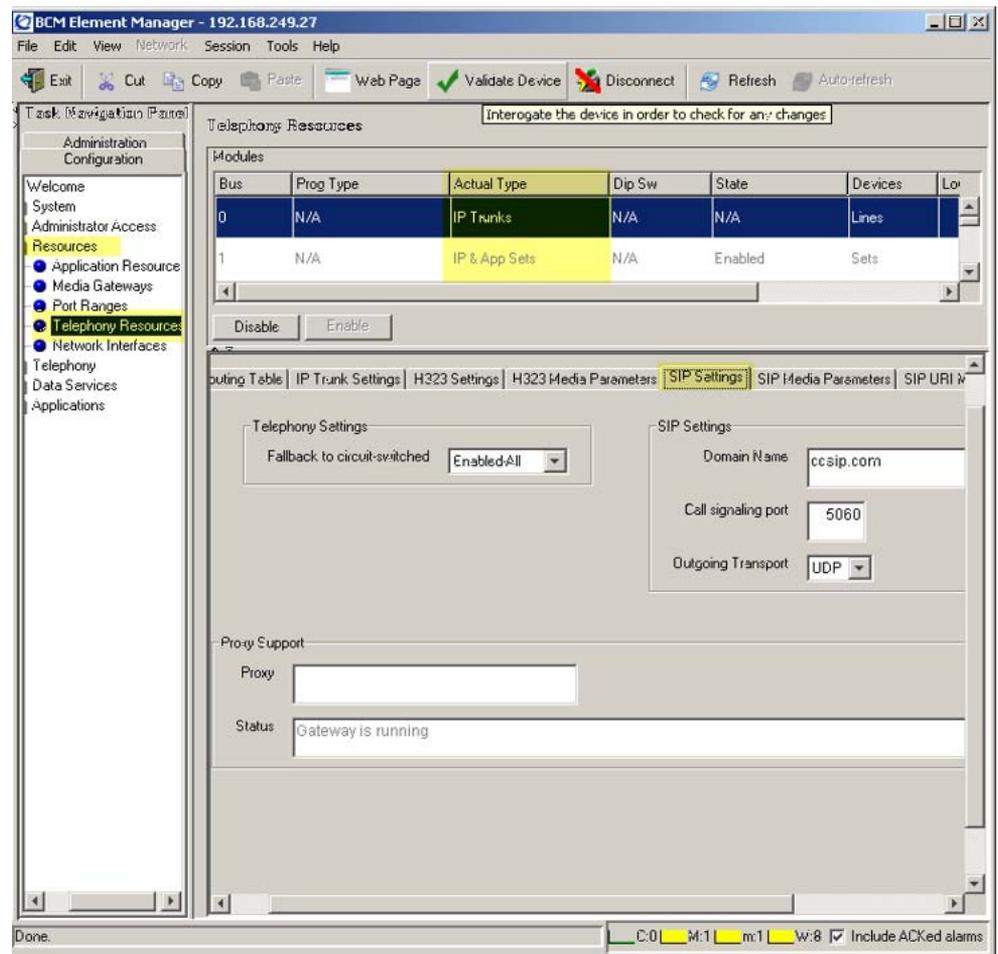
Table 11
H.323 Call Signaling Settings fields

Field	Value	Description
Call Signaling	Direct	Call signaling information is passed directly between H.323 endpoints. You must set up remote Gateways.
	Gatekeeper Resolved	All call signaling occurs directly between H.323 endpoints. This means that the Gatekeeper resolves the phone numbers into IP addresses, but the Gatekeeper is not involved in call signaling.
	Gatekeeper Routed	Gatekeeper Routed uses a Gatekeeper for call setup and control. In this method, call signaling is directed through the Gatekeeper.
	Gatekeeper Routed no RAS	Use this setting for a NetCentrex Gatekeeper. With this setting, the system routes all calls through the Gatekeeper but does not use any of the Gatekeeper Registration and Admission Services (RAS). Choose this option if RAS is not enabled on the NRS.
Call Signaling Port	<port value>	If VoIP applications are installed that require nonstandard call signaling ports, enter the port number here. Port number 0 means that the system uses the first available port. The default port for call signaling is 1720.
RAS Port	<port value>	If the VoIP application requires a nonstandard RAS port, enter the port number here. Port number 0 means that the system uses the first available port.
Enable H245 tunneling	<check box>	Select this field to allow H.245 messages within H.225. Restart the VoIP service for this feature to take effect.
Primary Gatekeeper IP	<IP address>	Fill in this field only if the network is controlled by a Gatekeeper. This is the IP address of the primary Gatekeeper (TLAN IP address).

Field	Value	Description
Backup Gatekeeper(s)	<IP address>	NetCentrex Gatekeeper does not support RAS. Any backup Gatekeepers must be entered in this field. Gatekeepers that use RAS can provide a list of backup Gatekeepers for the endpoint to use in the event of a primary Gatekeeper failure.
Alias names	NAME:<alias name>	Enter the alias names of the BCM required to direct call signals to your system. Note: The Alias name is case sensitive. It must match the name configured in NRS.
Registration TTL(s)	<numeric value>	Specifies the keep-alive interval.

- For SIP trunks, select the **SIP Settings** tab. See SIP Settings.

Figure 54
SIP Settings



- 11 Enter the information that supports your system. Refer to SIP Settings fields for more information.

Table 12
SIP Settings fields

Field	Value	Description
Fallback to Circuit-Switched	Disabled	Defines how you want the system to handle calls that the system fails to send over the VoIP trunk.
	Enabled-TDM	
	Enabled-All	Enabled-TDM enables fallback for calls originating on digital telephones. This is useful if your IP telephones are connected remotely, on the public side of the BCM network, because PSTN fallback is unlikely to result in better quality of service.
Domain Name		Type the domain name of the SIP network.
Call Signaling Port	<port value>	If VoIP applications are installed that require nonstandard call signaling ports, enter the port number here. Port number 0 means that the system uses the first available port.
Outgoing Transport	UDP	
	TCP	
Proxy		If entered, all SIP calls originate to this address.
Status	Read Only	This field displays the current status of the Gatekeeper.

—End—

Configuring VoIP lines

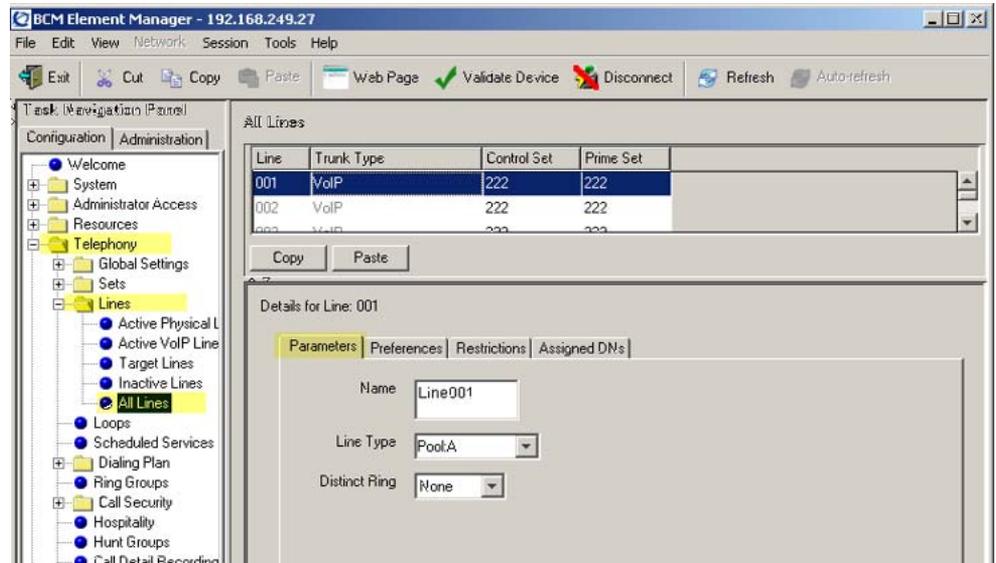
Voice over IP (VoIP) lines simulate traditional Central Office (CO) lines. VoIP lines transmit data over an IP network rather than over physical lines.

Configuring VoIP lines

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select Telephony > Lines > All Lines .
4	Highlight the individual line you wish to configure.
5	Select the Parameters tab.

See VoIP lines.

Figure 55
VoIP lines



- 6 Configure the Parameters tab appropriately for your network. Refer to VoIP line descriptions for configuration information.

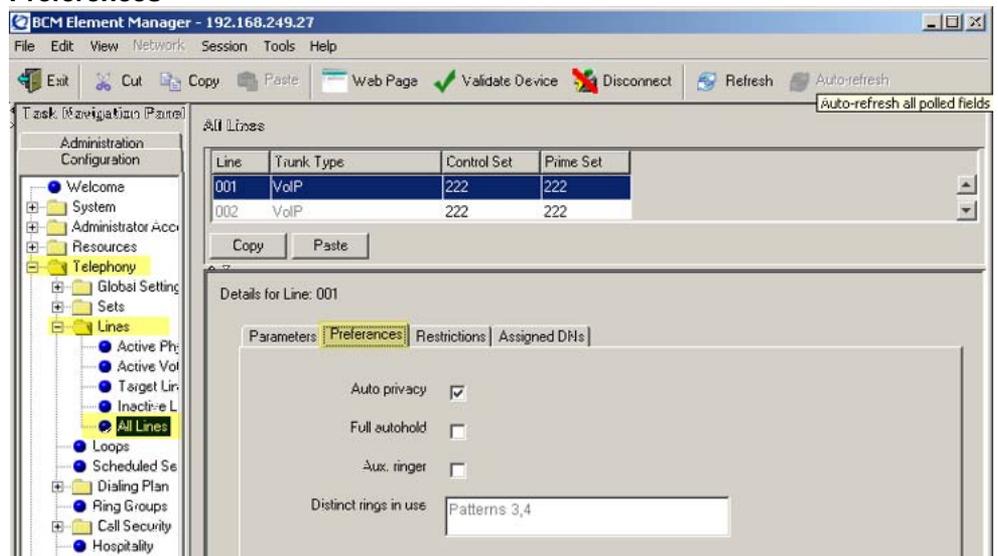
Table 13
VoIP line descriptions

Field	Value	Description
Line	001-060	Unique line number.
Trunk Type	VoIP	Ensure that the trunk type is set to VoIP when configuring VoIP lines.
Control Set		Identify a DN if you are using this line with scheduling. To change the DN, double-click the Control Set DN. For VoIP trunks, it is recommended that the Control Set be set to None because these are virtual trunks. Ensure that the VoIP trunk is assigned to a line pool.
Prime Set		Use the Prime Set if you want the line to be answered at another telephone when the line is not answered at the target telephone. To change the Prime set, double-click the Prime set DN. For VoIP trunks, it is recommended that the Prime Set be set to None because these are virtual trunks. Ensure that the VoIP trunk is assigned to a line pool.

Field	Value	Description
Name		Identify the line in a meaningful way.
Line Type	<p>Public</p> <p>DN:*</p> <p>Pool [A to O]</p>	<p>Defines how the line is used in relation to other lines in the system.</p> <p>If the line is to be shared among telephones, set to Public.</p> <p>If the line is assigned to only one telephone, set to DN:*</p> <p>If you are using routing, put the line into line pool (A to F).</p> <p>If you are using line pools, configure the target lines. If your system uses both H.323 and SIP trunks, assign H.323 trunks to one pool and SIP trunks to another.</p>
Distinct Ring	2, 3, 4, or None	For trunks assigned to line pools, set the Distinct Ring pattern to None.

- 7 Select the **Preferences** tab.
See Preferences.

Figure 56
Preferences



- 8 Configure the Preferences tab appropriately for your network.

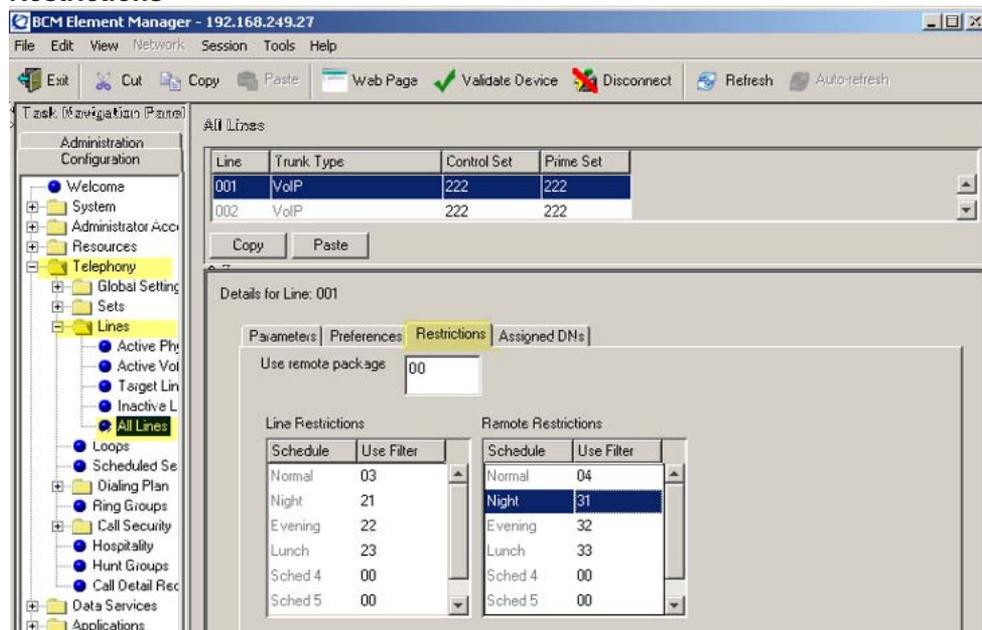
Refer to Preferences fields for configuration information.

Table 14
Preferences fields

Field	Value	Description
Auto Privacy	<check box>	Defines whether one BCM user can select a line in use at another telephone to join an existing call. For more information, see <i>BCM 4.0 Device Configuration Guide</i> (N0060600).
Full autohold	<check box>	Enables or disables Full autohold. When enabled, if a caller selects an idle line but does not dial any digits, that line is automatically placed on hold if the caller selects another line. Change the default setting only if Full autohold is required for a specific application.
Aux. ringer	<check box>	If your system is equipped with an external ringer, you can enable this setting so that this line rings at the external ringer.
Distinct rings in use	Read only	Indicates whether a special ring is assigned.

- 9 Select the **Restrictions** tab.
See Restrictions.

Figure 57
Restrictions



- 10 Configure the Restrictions tab appropriately for your network.

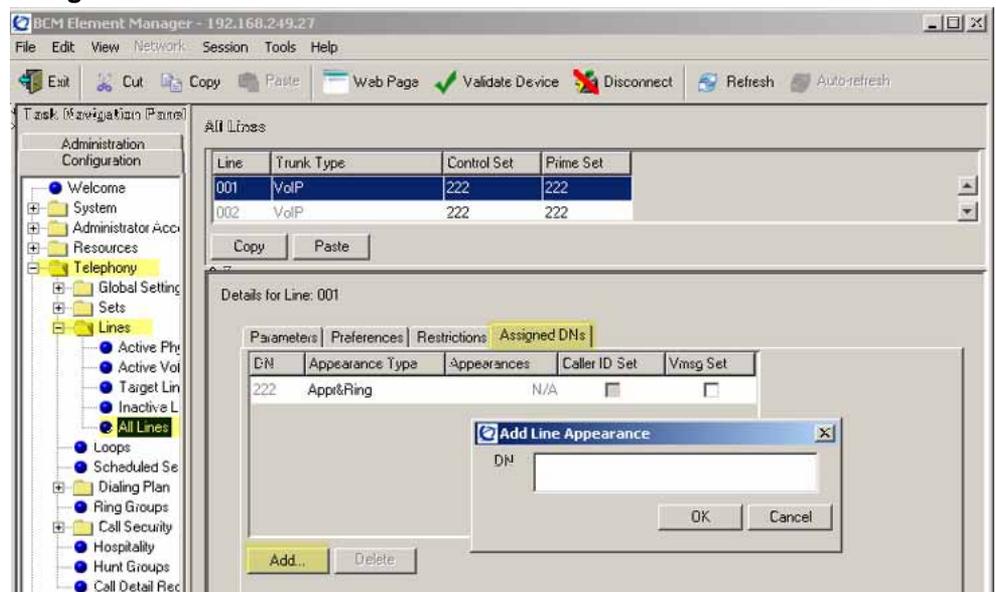
Refer to Restrictions fields for configuration information.

Table 15
Restrictions fields

Field	Value	Description
Use remote package	< package #>	If the line is used to receive external calls or calls from other nodes on the private network, ensure that you indicate a remote package that provides only the availability that you want for external callers. This attribute is typically used for tandeming calls.
Schedule	Default: Normal, Night, Evening, Lunch, Sched 4, Sched 5, Sched 6	
Line Restrictions - Use Filter	<00-99>	Enter the restriction filter number that applies to each schedule. These settings control outgoing calls.
Remote Restrictions - Use Filter	<00-99>	Enter the restriction filter that applies to each schedule. These settings provide call controls for incoming calls over a private network or from a remote user dialing in over PSTN.

- 11 Select the **Assigned DNs** tab.
See Assigned DNs.

Figure 58
Assigned DNs



- 12 Edit the listed DNs or click Add to add a DN as required.
- 13 Enter the appropriate information for your network. Refer to Assigned DNs fields for configuration information.

Table 16
Assigned DNs fields

Field	Value	Description
DN		Unique number
Appearance Type	Ring Only Appr&Ring Appr Only	Select Appr Only or Appr&Ring if the telephone has an available button. Otherwise select Ring Only.
Appearances		Target lines can have more than one appearance to accommodate multiple calls. For telephones that have these lines set to Ring Only, set to None.
Caller ID Set	<check box>	When enabled, displays caller ID for calls coming in over the target line.
Vmsg Set	<check box>	When enabled, an indicator appears on the telephone when a message is waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

—End—

Configuring target lines

Target lines are virtual communication paths between trunks and telephones on the BCM system. They are incoming lines only and cannot be selected for outgoing calls or networking applications.

Configuring target lines

Step	Action
------	--------

- | | |
|---|--|
| 1 | Log on to Element Manager. |
| 2 | In the Task Navigation Panel , select the Configuration tab. |
| 3 | Select Telephony > Lines > Target Lines . |
| 4 | Highlight the individual line you wish to configure. |

- Select the **Parameters** tab and enter the appropriate information for your network.
See Parameters. Refer to Parameters fields for configuration information.

Figure 59
Parameters

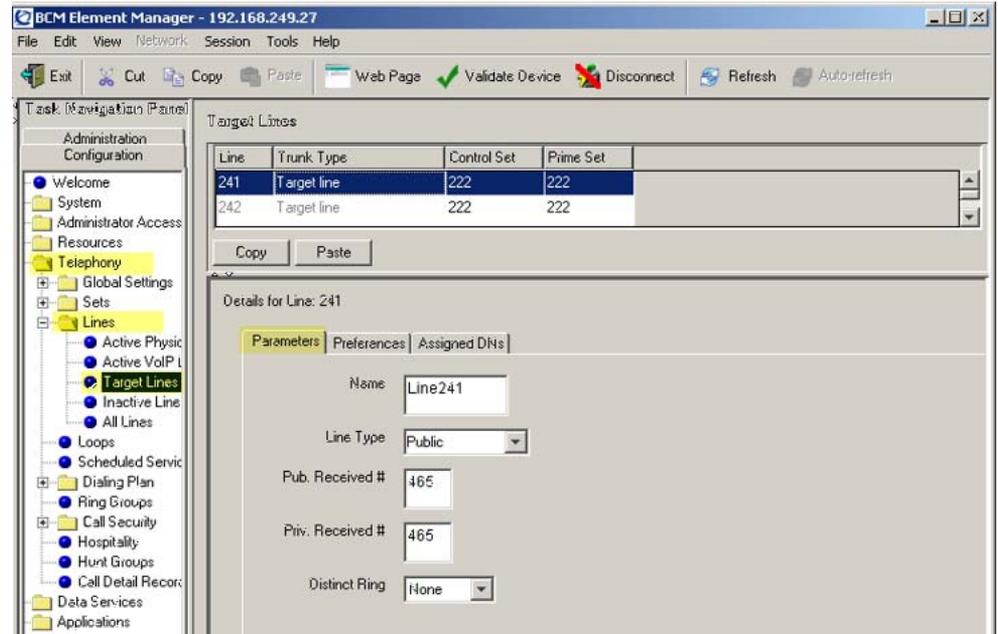


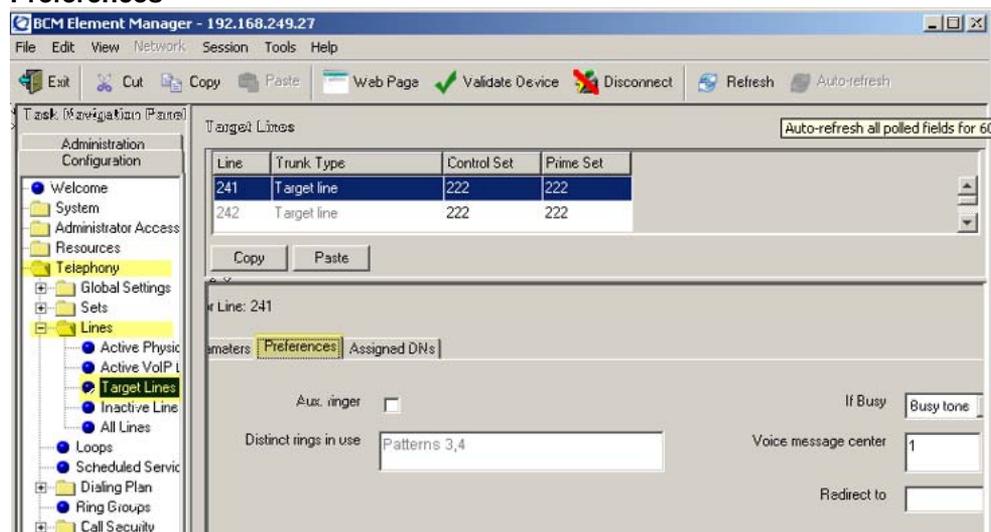
Table 17
Parameters fields

Field	Value	Description
Line Type	Public DN:*	If the line is to be shared among telephones, select Public. If the line is only assigned to one telephone, select DN:*
Pub. Received #		Confirm the existing number or enter a public received number (PSTN DID or PRI trunks) that the system uses to identify calls from the public network to the target line. The public received number cannot be the same as the beginning digits of a line pool access code or destination code.

Field	Value	Description
Priv. Received #		If private network trunks (PRI or VoIP trunks) are configured, enter a private received number. The private received number specifies the digits the system uses to identify calls from the private network to a target line. This number is usually the same as the DN.
Distinct Ring	2, 3, 4, or None	If you want this line to have a special ring, select a ring pattern.

- 6 Select the **Preferences** tab and enter the appropriate information for your network.
See Preferences. Refer to Preferences fields for configuration information.

**Figure 60
Preferences**



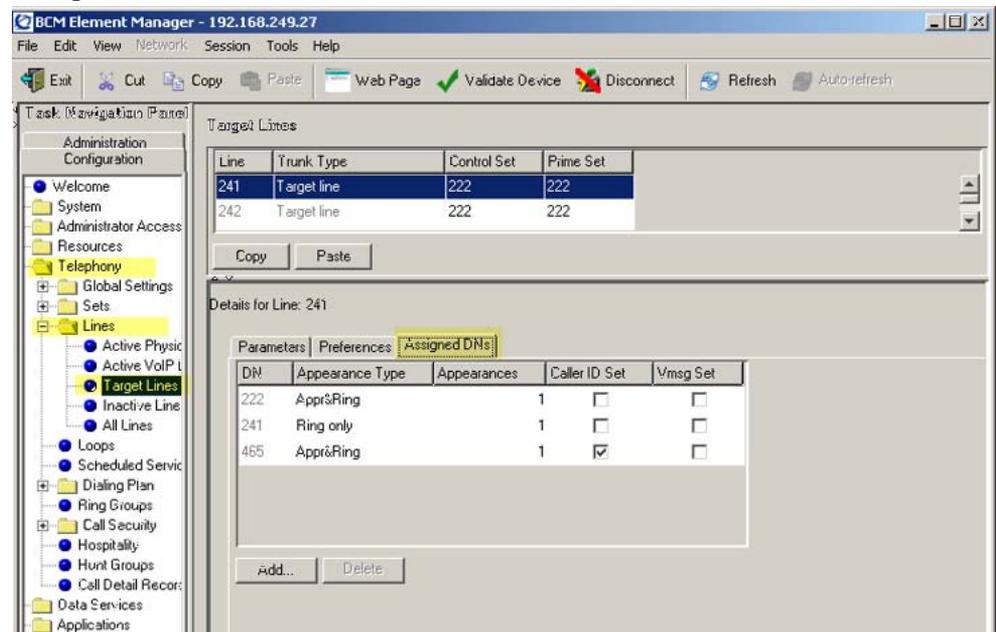
**Table 18
Preferences fields**

Field	Value	Description
Aux. Ringer	<check box>	If your system is equipped with an external ringer, you can enable this setting so that this line rings at the external ringer.
If Busy	Busy tone To Prime	To automatically direct calls to the prime telephone, select To prime. Otherwise, select Busy tone.
Distinct rings in use	Read only	

Field	Value	Description
Voice message center		If the system is using a remote voice mail, select the center configured with the contact number.
Redirect to		To automatically direct calls out of the system to a specific telephone, such as a head office answer attendant, enter that remote number here. Ensure that you include the proper routing information.

- 7 Select the **Assigned DNs** tab.
See Assigned DNs.

Figure 61
Assigned DNs



- 8 Edit the listed DNs, or click **Add** to add a DN as required.
- 9 Enter the appropriate information for your network.
Refer to Assigned DNs fields for configuration information.

—End—

BCM50 configuration

This chapter describes configuration procedures for the Business Communications Manager 50 (BCM50) system.

Element Manager as viewed on your system may differ slightly from the screens shown in this chapter because you can customize the column display in Element Manager.

BCM50 configuration procedures

The sequence of BCM50 configuration procedures is as follows:

- Configuring incoming VoIP trunks
- Verifying system license and keycodes
- Configuring VoIP trunk media parameters
- Configuring local Gateway parameters
- Configuring VoIP lines
- Configuring target lines

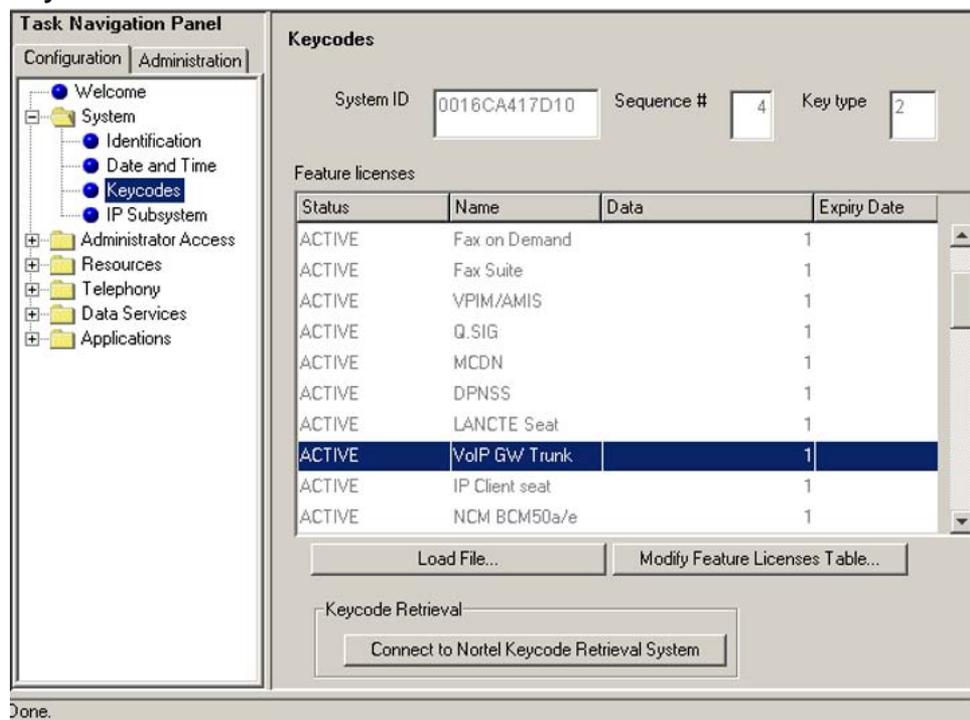
Configuring incoming VoIP trunks

Perform the following procedure to configure incoming VoIP trunks.

Configuring incoming VoIP trunks

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select System > Keycodes . See Keycodes.

Figure 62
Keycodes



- 4 Load new Keycodes by loading a new keycode file or connecting to Nortel's Keycode Retrieval System (KRS). For more information about keycodes and keycode retrieval, see *Keycode Installation Guide* (NN40010-301).

—End—

Verifying system license and keycodes

Perform the following procedure to verify system license and keycodes.

Verifying system license and keycodes

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select System > Keycodes . See Keycodes.

- 4 In the **Name** column, scroll down to **VoIP GW Trunk**. The number of license keys you have are listed in the Data column.

—End—

Configuring VoIP trunk media parameters

Perform the following procedure to configure VoIP trunk media parameters.

Configuring VoIP trunk media parameters

Step	Action
------	--------

- 1 Log on to Element Manager.
- 2 In the **Task Navigation Panel**, select the **Configuration** tab.
- 3 Select **Resources > Telephony Resources**.
See Telephony Resources.

Figure 63
Telephony Resources

The screenshot shows the 'Telephony Resources' configuration page. On the left is the 'Task Navigation Panel' with a tree view containing 'System', 'Administrator Access', 'Resources', 'Application Resources', 'Media Gateways', 'Port Ranges', 'Telephony Resources' (selected), 'Dial Up Interfaces', 'Telephony', 'Data Services', and 'Applications'. The main area is titled 'Telephony Resources' and contains a 'Modules' table.

Location	Module type	Bus	State	Devices	Low	High	Total	Busy
Internal	IP & Application Sets	1	N/A	Sets	N/A	N/A	0	0
Internal	IP Trunks	N/A/N/A	Lines		1	12	12	0
Internal	BRI Loop	3	Enabled	Lines	61	64	4	0
Internal	Sets	4	Enabled	Sets	N/A	N/A	0	0

Below the table are 'Disable' and 'Enable' buttons. The 'Details for Module: Internal' section is expanded, showing tabs for 'Routing Table', 'H.323 Settings', 'H.323 Media Parameters' (selected), 'SIP Settings', 'SIP Media Parameters', and 'SIP URI Map'. The 'H.323 Media Parameters' tab contains 'Preferred Codecs' and 'Settings' sections.

Preferred Codecs:

Codec Preferences:

- Available list: (empty)
- Selected list: G.729, G.723, G.711-uLaw, G.711-aLaw

Settings:

- Enable Voice Activity Detection:
- Jitter buffer: Auto
- G.729 payload size (ms): 30
- G.723 payload size (ms): 30
- G.711 payload size (ms): 30
- Incremental payload size:
- Enable T.38 fax:
- Force G.711 for 3.1k audio:

- 4 In the **Modules** panel, select the line where the **Actual Type** column is set to **IP Trunks**.
- 5 Select the **H.323 Media Parameters** or **SIP Media Parameters** tab.

- 6 Enter the information that supports your system. Ensure that these settings are consistent with the other systems on your network. Refer to H.323 Media Parameters fields and SIP Media Parameters fields for a description of the parameters.

—End—

Table 19
H.323 Media Parameters fields

Field	Value	Description
Preferred Codecs	G.711 -uLaw G.711 -aLaw G.729 G.723	Add codecs to the Selected list and order them in the order in which you want the system to attempt to use them. The system attempts to use the codecs in top-to-bottom sequence. Performance note: Codecs on all networked BCMs must be consistent to ensure the proper functionality of interacting features such as Transfer and Conference. Systems running BCM Release 3.5 or later allow codec negotiation and renegotiation to accommodate inconsistencies in codec settings over VoIP trunks.
Enable Voice Activity Detection	<check box>	Voice Activity Detection (VAD), also known as silence suppression, identifies periods of silence in a conversation and stops sending IP speech packets during those periods. In a typical telephone conversation, most of the conversation is half-duplex, meaning that one person is speaking while the other is listening. If VAD is enabled, no voice packets are sent from the listener end. This greatly reduces bandwidth requirements. G.723.1 and G.729 support VAD. G.711 does not support VAD. Performance note: VAD on all networked BCMs and IPT systems must be consistent to ensure functionality of features such as Transfer and Conference. The Payload size on the IPT must be set to 30ms.

Field	Value	Description
Jitter Buffer	Auto None Small Medium Large	Select the size of jitter buffer for your system.
G.729 payload size (ms) G.723 payload size (ms) G.711 payload size (ms)	10,20,30,40,50,60 30 10,20,30,40,50,60	Set the maximum required payload size, per codec, for the VoIP calls sent over H.323 trunks. Note: Payload size can also be set for Nortel IP telephones. See <i>BCM 4.0 Telephony Device Installation Guide</i> (N0060609).
Incremental payload size	<check box>	When enabled, the system advertises a variable payload size (40, 30, 20, 10 ms).
Enable T.38 fax	<check box>	When enabled, the system supports T.38 fax over IP. Caution: Fax tones broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of the fax machine. To minimize the possibility of your VoIP calls being dropped due to fax tone interference: <ul style="list-style-type: none"> place the fax machine away from other telephones turn the fax machine's speaker volume to the lowest level, or off, if available
Force G.711 for 3.1k Audio	<check box>	When enabled, the system forces the VoIP trunk to use the G.711 codec for 3.1k audio signals, such as modem or TTY machines. Note: You also can use this setting for fax machines if T.38 fax is not enabled on the trunk.

Table 20
SIP Media Parameters fields

Field	Value	Description
Preferred Codecs	G.711 -uLaw G.711 -aLaw G.729	Add codecs to the Selected list and order them in the order in which you want the system to attempt to use them. The system attempts to use the codecs in a top-to-bottom sequence.

Field	Value	Description
	G.723	<p>Performance note: Codecs on all networked BCMs must be consistent to ensure the proper functionality of interacting features such as Transfer and Conference.</p> <p>Systems running BCM Release 3.5 or later allow codec negotiation and renegotiation to accommodate inconsistencies in codec settings over VoIP trunks.</p>
Enable Voice Activity Detection	<check box>	<p>Voice Activity Detection (VAD), also known as silence suppression, identifies periods of silence in a conversation and stops sending IP speech packets during those periods. In a typical telephone conversation, most of the conversation is half-duplex, meaning that one person is speaking while the other is listening. If VAD is enabled, no voice packets are sent from the listener end. This greatly reduces bandwidth requirements. G.723.1 and G.729 support VAD. G.711 does not support VAD.</p> <p>Performance note: VAD on all networked BCMs and IPT systems must be consistent to ensure functionality of features such as Transfer and Conference. The Payload size on the IPT must be set to 30ms.</p>
Jitter Buffer	Auto None Small Medium Large	Select the size of jitter buffer for your system.
G.729 payload size (ms) G.723 payload size (ms) G.711 payload size (ms)	10,20,30,40,50,60 30 10,20,30,40,50,60	<p>Set the maximum required payload size, per codec, for the VoIP calls sent over H.323 trunks.</p> <p>Note: Payload size can also be set for Nortel IP telephones. See <i>BCM 4.0 Telephony Device Installation Guide</i> (N0060609).</p>
Enable T.38 fax	<check box>	<p>When enabled, the system supports T.38 fax over IP.</p> <p>Caution: Fax tones broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of</p>

Field	Value	Description
		<p>the fax machine. To minimize the possibility of your VoIP calls being dropped due to fax tone interference:</p> <ul style="list-style-type: none"> • place the fax machine away from other telephones • turn the fax machine's speaker volume to the lowest level, or off, if available

Configuring local Gateway parameters

Perform the following procedure to configure local Gateway parameters.

Configuring local Gateway parameters

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	In the Module Panel , select the line in which the Actual Type column is set to IP Trunks . See Telephony Resources.
4	Select the IP Trunk Settings tab and enter the information that supports your system. See IP Trunk Settings. Refer to IP Trunk Settings fields for information about the IP Trunk Settings fields.

Figure 64
IP Trunk Settings

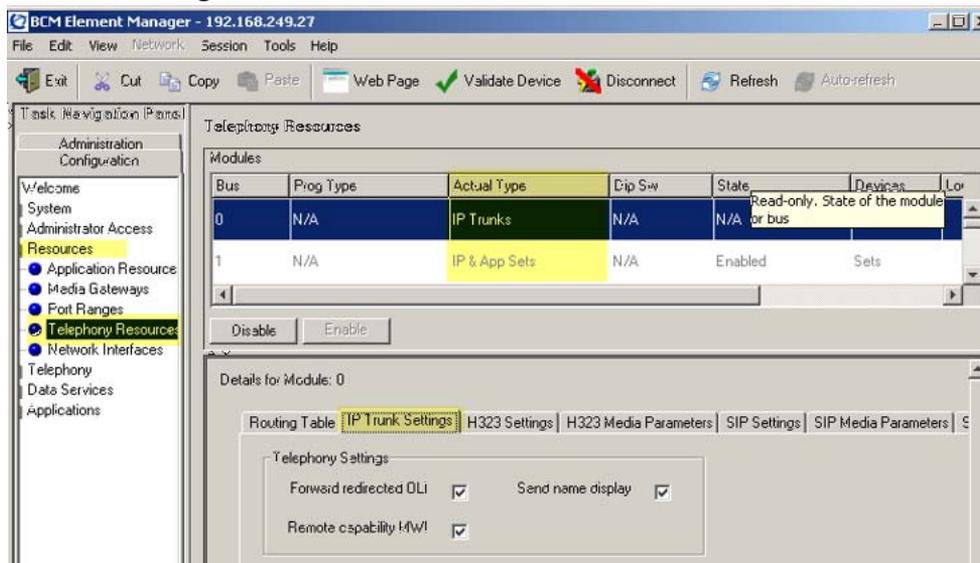
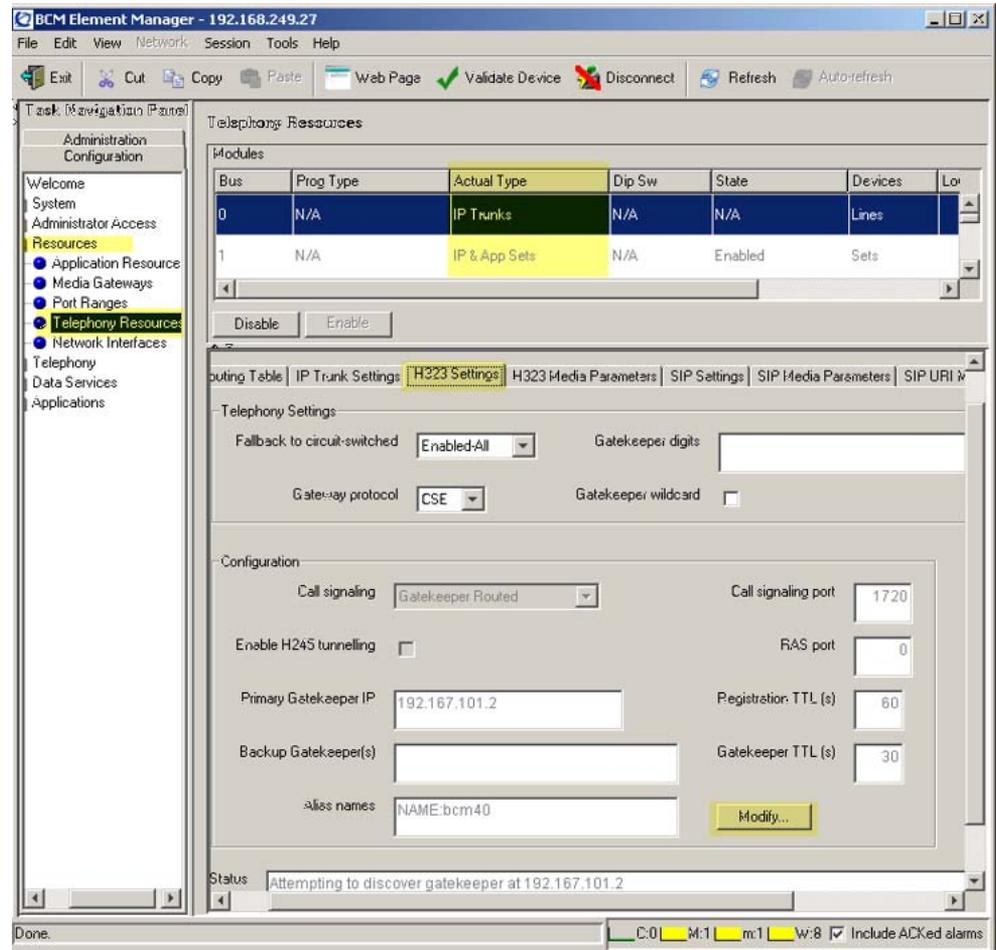


Table 21
IP Trunk Settings fields

Field	Value	Description
Forward redirected OLI	<check box>	If enabled, the OLI of an internal telephone is forwarded over the VoIP trunk when a call is transferred to an external number over the private VoIP network. If disabled, only the CLID of the transferred call is forwarded.
Send name display	<check box>	If enabled, the telephone name is sent with outgoing calls to the network.
Remote capability MWI	<check box>	This setting must coordinate with the functionality of the remote system hosting remote voice mail.

- 5 For H.323 VoIP trunks, select the **H.323 Settings** tab. See H.323 Settings.

Figure 65
H.323 Settings



- 6 When implementing your dialing plan, in the **H.323 Settings** tab, select a value for **Fall back to circuit-switched**. This determines how the system handles calls if the IP network cannot be used.
- 7 For **Gateway protocol**, select **CSE**.
- 8 Scroll down to **Alias Name** and click **Modify**. The Modify Call Signaling Settings page appears.
- 9 Enter the information that supports your system. Applying the changes made to the Call Signaling Settings causes all H.323 calls to be dropped. It is recommended that you make changes to the Call Signaling Settings during off-peak hours or a scheduled maintenance window.

Refer to H.323 Call Signaling Settings fields.

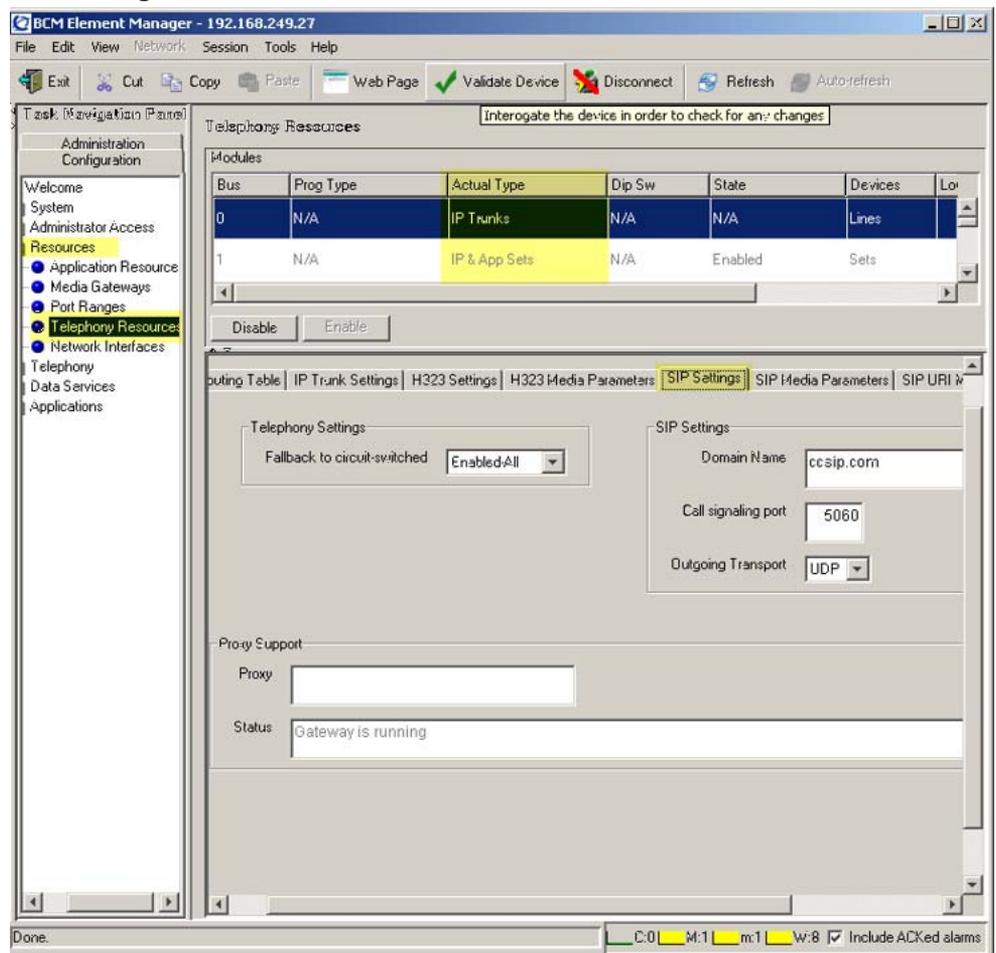
Table 22
H.323 Call Signaling Settings fields

Field	Value	Description
Call Signaling	Direct	Call signaling information is passed directly between H.323 endpoints. You must set up remote Gateways.
	Gatekeeper Resolved	All call signaling occurs directly between H.323 endpoints. This means that the Gatekeeper resolves the phone numbers into IP addresses, but the Gatekeeper is not involved in call signaling.
	Gatekeeper Routed	Gatekeeper Routed uses a Gatekeeper for call setup and control. In this method, call signaling is directed through the Gatekeeper.
	Gatekeeper Routed no RAS	Use this setting for a NetCentrex Gatekeeper. With this setting, the system routes all calls through the Gatekeeper but does not use any of the Gatekeeper Registration and Admission Services (RAS). Choose this option if RAS is not enabled on the NRS.
Call Signaling Port	<port value>	If VoIP applications are installed that require nonstandard call signaling ports, enter the port number here. Port number 0 means that the system uses the first available port. The default port for call signaling is 1720.
RAS Port	<port value>	If the VoIP application requires a nonstandard RAS port, enter the port number here. Port number 0 means that the system uses the first available port.
Enable H245 tunneling	<check box>	Select this field to allow H.245 messages within H.225. Restart the VoIP service for this feature to take effect.
Primary Gatekeeper IP	<IP address>	Fill in this field only if the network is controlled by a Gatekeeper. This is the IP address of the primary Gatekeeper (TLAN IP address).
Backup Gatekeeper(s)	<IP address>	NetCentrex Gatekeeper does not support RAS. Any backup Gatekeepers must be entered in this field. Gatekeepers that use RAS can provide a list of backup Gatekeepers for the endpoint to use in the event of a primary Gatekeeper failure.

Field	Value	Description
Alias names	NAME:<alias name>	Enter the alias names of the BCM required to direct call signals to your system. Note: The Alias name is case sensitive. It must match the name configured in NRS.
Registration TTL(s)	<numeric value>	Specifies the keep-alive interval.

- 10 For SIP trunks, select the **SIP Settings** tab. See SIP Settings.

Figure 66
SIP Settings



- 11 Enter the information that supports your system.

Refer to SIP Settings fields for more information.

Table 23
SIP Settings fields

Field	Value	Description
Fallback to Circuit-Switched	Disabled	Defines how you want the system to handle calls that the system fails to send over the VoIP trunk.
	Enabled-TDM	
	Enabled-All	Enabled-TDM enables fallback for calls originating on digital telephones. This is useful if your IP telephones are connected remotely, on the public side of the BCM network, because PSTN fallback is unlikely to result in better quality of service.
Domain Name		Type the domain name of the SIP network.
Call Signaling Port	<port value>	If VoIP applications are installed that require nonstandard call signaling ports, enter the port number here. Port number 0 means that the system uses the first available port.
Outgoing Transport	UDP	
	TCP	
Proxy		If entered, all SIP calls originate to this address.
Status	Read Only	This field displays the current status of the Gatekeeper.

—End—

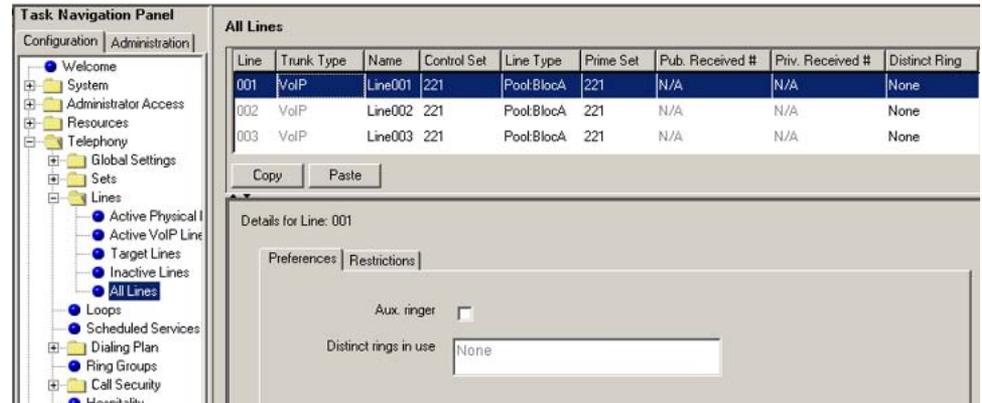
Configuring VoIP lines

Voice over IP (VoIP) lines simulate traditional Central Office (CO) lines. VoIP lines transmit data over an IP network rather than over physical lines.

Configuring VoIP lines

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select Telephony > Lines > All Lines .
4	Highlight the individual line you wish to configure.
5	Select the Preferences tab. See Preferences.

Figure 67
Preferences



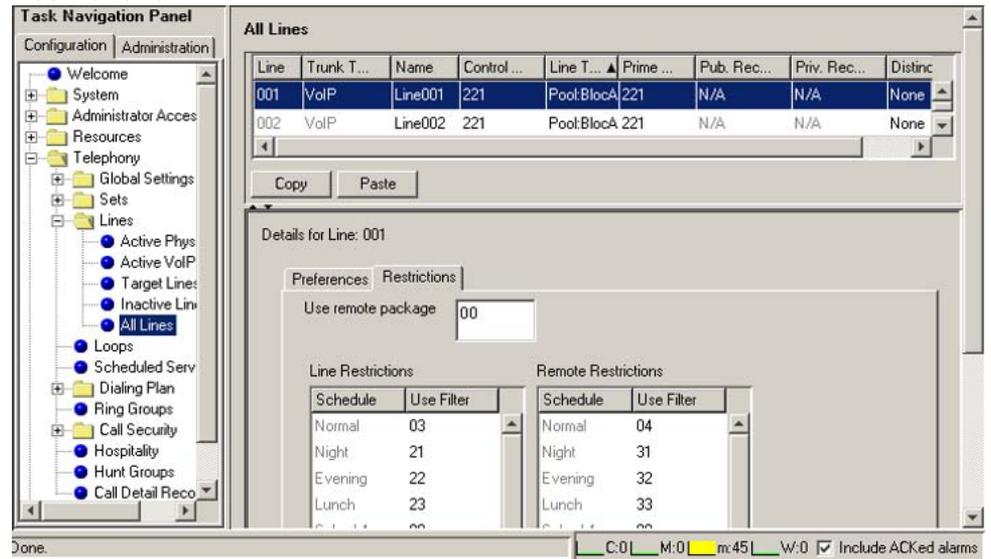
- 6 Configure the Preferences tab appropriately for your network. Refer to Preferences fields for configuration information.

Table 24
Preferences fields

Field	Value	Description
Aux. ringer	<check box>	If your system is equipped with an external ringer, you can enable this setting so that this line rings at the external ringer.
Distinct rings in use	Read only	Indicates whether a special ring is assigned.

- 7 Select the **Restrictions** tab. See Restrictions.

Figure 68
Restrictions



- 8 Configure the Restrictions tab appropriately for your network. Refer to Restrictions fields for configuration information.

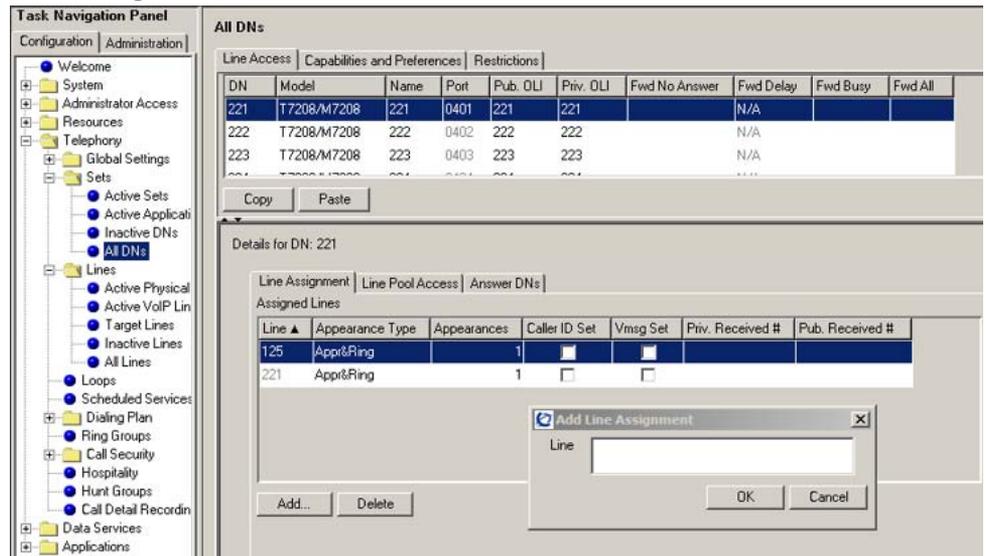
Table 25
Restrictions fields

Field	Value	Description
Use remote package	< package #>	If the line is used to receive external calls or calls from other nodes on the private network, ensure that you indicate a remote package that provides only the availability that you want for external callers. This attribute is typically used for tandeming calls.
Schedule	Default: Normal, Night, Evening, Lunch, Sched 4, Sched 5, Sched 6	
Line Restrictions - Use Filter	<00-99>	Enter the restriction filter number that applies to each schedule. These settings control outgoing calls.
Remote Restrictions - Use Filter	<00-99>	Enter the restriction filter that applies to each schedule. These settings provide call controls for incoming calls over a private network or from a remote user dialing in over PSTN.

- 9 In the **Task Navigation Panel**, in the **Configuration** tab, select **Telephony > Lines > All Lines**.

- 10 Highlight the individual line you wish to configure.
- 11 Select the **Line Assignment** tab. See Line Assignment.

Figure 69
Line Assignment



- 12 Edit the listed DN's, or click **Add** to add a DN as required.
- 13 Enter the appropriate information for your network. Refer to Assigned DN's fields for configuration information.

Table 26
Assigned DN's fields

Field	Value	Description
DN		Unique number
Appearance Type	Ring Only Appr&Ring Appr Only	Select Appr Only or Appr&Ring if the telephone has an available button. Otherwise select Ring Only.
Appearances		Target lines can have more than one appearance to accommodate multiple calls. For telephones that have these lines set to Ring Only, set to None.

Field	Value	Description
Caller ID Set	<check box>	When enabled, displays caller ID for calls coming in over the target line.
Vmsg Set	<check box>	When enabled, an indicator appears on the telephone when a message is waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

—End—

Configuring target lines

Target lines are virtual communication paths between trunks and telephones on the BCM system. They are incoming lines only and cannot be selected for outgoing calls or networking applications.

Configuring target lines

Step	Action
1	Log on to Element Manager.
2	In the Task Navigation Panel , select the Configuration tab.
3	Select Telephony > Lines > Target Lines .
4	Highlight the individual line you wish to configure.
5	Select the Preferences tab and enter the appropriate information for your network. See Preferences. Refer to Preferences fields for configuration information.

Figure 70
Preferences

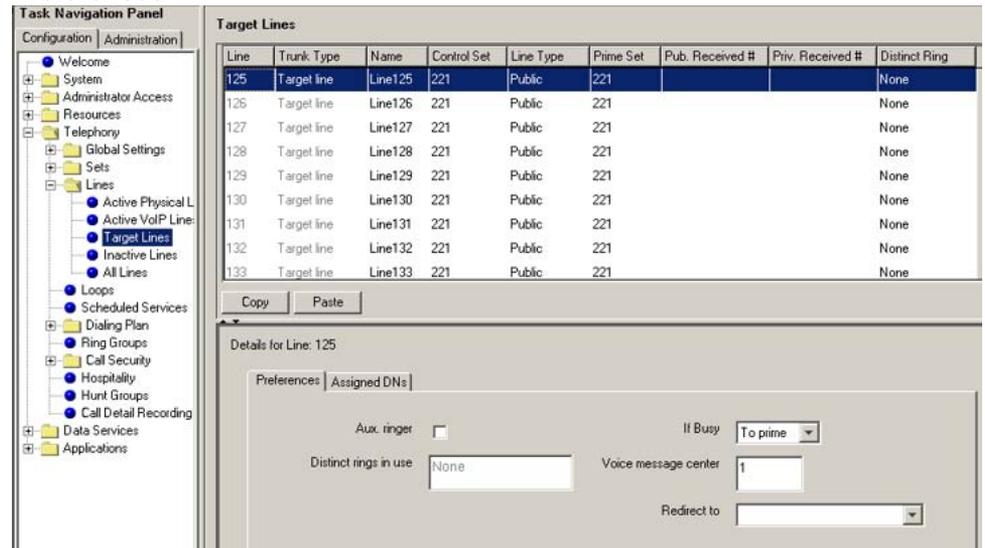


Table 27
Preferences fields

Field	Value	Description
Aux. Ringer	<check box>	If your system is equipped with an external ringer, you can enable this setting so that this line rings at the external ringer.
If Busy	Busy tone To Prime	To automatically direct calls to the prime telephone, select To prime. Otherwise, select Busy tone.
Distinct rings in use	Read only	
Voice message center		If the system is using a remote voice mail, select the center configured with the contact number.
Redirect to		To automatically direct calls out of the system to a specific telephone, such as a head office answer attendant, enter that remote number here. Ensure that you include the proper routing information.

- 6 Select the **Assigned DN's** tab.
See Assigned DN's.

Figure 71
Assigned DNs

The screenshot shows the BCM50 configuration interface. On the left is the 'Task Navigation Panel' with a tree view containing folders like System, Administrator Access, Resources, Telephony, Global Settings, Sets, Lines, Loops, Scheduled Services, Dialing Plan, Ring Groups, Call Security, Hospitality, Hunt Groups, Call Detail Recording, Data Services, and Applications. The 'Lines' folder is expanded, showing 'Active Physical L', 'Active VoIP Line', 'Target Lines', 'Inactive Lines', and 'All Lines'. The 'Target Lines' folder is selected.

The main area displays a 'Target Lines' table:

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #	Distinct Ring
125	Target line	Line125	221	Public	221			None
126	Target line	Line126	221	Public	221			None
127	Target line	Line127	221	Public	221			None
128	Target line	Line128	221	Public	221			None
129	Target line	Line129	221	Public	221			None
130	Target line	Line130	221	Public	221			None
131	Target line	Line131	221	Public	221			None
132	Target line	Line132	221	Public	221			None
133	Target line	Line133	221	Public	221			None

Below the table are 'Copy' and 'Paste' buttons. The 'Details for Line: 125' section has tabs for 'Preferences' and 'Assigned DNs'. The 'Assigned DNs' tab shows a table with columns: DN, Appearance Type, Appearances, Caller ID Set, and Vmsg Set. The table contains one row: DN: 221, Appearance Type: Appri&Ring, Appearances: 1, Caller ID Set: , Vmsg Set: . Below this table are 'Add...' and 'Delete' buttons. An 'Add Line Appearance' dialog box is open, showing a 'DN' input field and 'OK' and 'Cancel' buttons.

- 7 Edit the listed DNs, or click **Add** to add a DN as required.
- 8 Enter the appropriate information for your network. Refer to Assigned DNs fields for configuration information.

—End—

Testing and troubleshooting

This chapter contains procedures to test and troubleshoot your Communication Server 1000/Business Communications Manager (BCM) integration.

The sections in this chapter are as follows:

- Testing
 - Testing the integration from the BCM system
 - Testing the integration from the CS 1000 system
- Troubleshooting
 - BCM is unable to contact the gatekeeper at IP address
 - Unable to complete any calls
 - Cannot make calls between the CS 1000 and BCM
 - BCM fails to register to NRS
 - H.323 Gateway service is down

Testing

The CS 1000/BCM integration is considered successful if BCM and Network Routing Service (NRS) are able to register to each other. You can determine this from either the CS 1000 side or the BCM side.

Successful call completion is not a criterion of integration success because call completion is dependent on the dialing plan and how it is implemented. For information about dialing plans, see *Dialing Plans: Description* (553-3001-183).

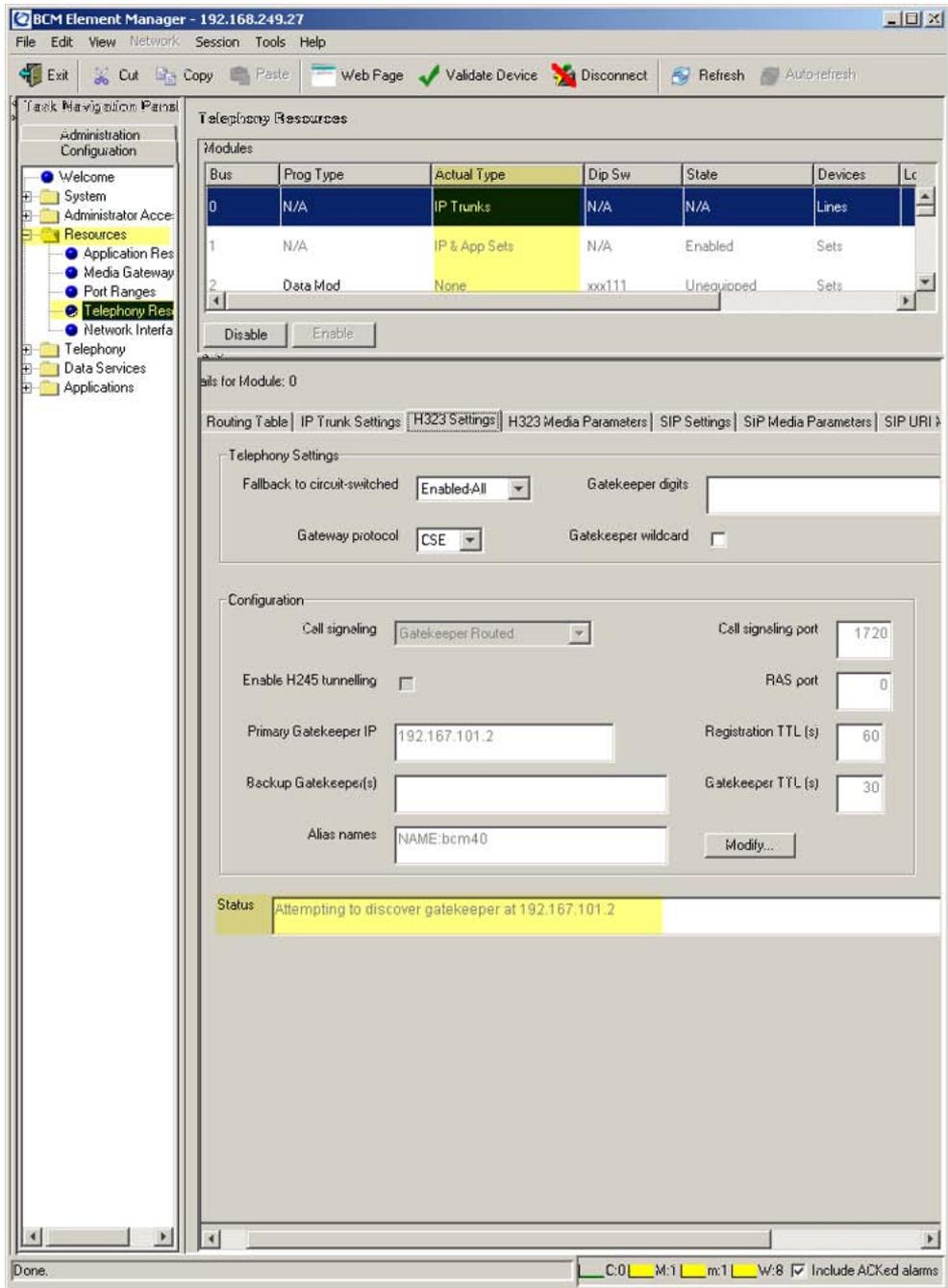
Testing the integration from the BCM system

Step	Action
------	--------

- | | |
|---|--|
| 1 | Log on to Element Manager on the BCM system. |
|---|--|

- 2 Select **Resources > Telephony Resources**.
See Status.

Figure 72
Status



- 3 In the **Actual Type** column, highlight **IP Trunks**.

- 4 In the bottom half of the page, select the **H323 Settings** tab.
- 5 Scroll down to the **Status** bar to determine if the two systems are successfully registered.

—End—

Testing the integration from the CS 1000 system

To determine if the two systems are registered from the CS 1000 side, check the status of the endpoints. Refer to the procedure Checking the status of registered endpoints.

Troubleshooting

Refer to these troubleshooting procedures to resolve common integration issues.

BCM is unable to contact the gatekeeper at IP address

Step	Action
1	Check whether you are able to ping the Gatekeeper across the network. If not, a routing issue can exist in your network. Contact your network administrator to resolve any routing issues.
2	Check that the correct Gateway endpoint IP address is configured in BCM. You may need to reset the feps service under the Service Manager.
3	Check that the correct Gateway endpoint IP address is configured in the CS 1000.
4	Check that the Alias name is properly configured in BCM. The alias name is case sensitive and must match exactly what is configured in the CS 1000.
5	Check that the Gateway protocol is set to CSE.

—End—

Unable to complete any calls

Step	Action
1	Check whether the BCM and Gatekeeper have established connectivity. If not, ensure that the BCM and NRS can communicate with each other.
2	Check that the line is configured for outgoing calls. DNs need to have lines configured for both incoming and outgoing calls. Check your networks dialing plan or see <i>Dialing Plans: Description</i> (553-3001-183).
3	Verify that the dialing plan has been properly implemented on both the CS 1000 and BCM. For more information about dialing plans, see <i>Dialing Plans: Description</i> (553-3001-183).
—End—	

Cannot make calls between the CS 1000 and BCM

Symptoms:

- calls between the CS 1000 and BCM fail
- CDP calls fail
- no channel/circuit is available

Step	Action
1	Verify your dialing plan and call routing.
2	On the BCM, log on to Element Manager and select Telephony > Dialing Plan > Private Network .
3	Verify that Private Network Type is set to CDP .
4	Ensure that packets are not blocked by your network firewall.
—End—	

BCM fails to register to NRS

Symptoms:

- BCM fails to register to the NRS

- calls fail between the CS 1000 and BCM in both directions

Step	Action
------	--------

- | | |
|---|--|
| 1 | Check whether you can ping the BCM from the NRS command line. If unsuccessful, check your network settings. Note that the NRS does not respond to pings. |
| 2 | In the H323 Settings tab for IP trunks under Resources > Telephony Resources , verify that the BCM includes its alias name as "NAME:aliasname". |
| 3 | Verify that the Alias names match on the NRS and BCM. |
| 4 | In the H323 Settings tab for IP trunks under Resources > Telephony Resources , make sure the Gateway protocol is set to CSE . |
| 5 | Verify that the NRS has the proper routing entries. |

—End—

H.323 Gateway service is down

Symptoms:

- VoIP H.323 Gateway service is down
- VoIP Gateway cannot be started manually
- VoIP Gateway does not start after a reboot or power cycle

Step	Action
------	--------

- | | |
|---|---|
| 1 | On the BCM, log on to Element Manager. |
| 2 | Select Resources > Telephony Resources . |
| 3 | In the Actual Type column, highlight VoIP Trunks . |
| 4 | Select the H.323 Settings tab and verify that the Call Signaling Port is set to 1720 . |
| 5 | Refer to the procedure Testing the integration from the BCM system. |

—End—

Enterprise: Common

Solution Integration Guide for Communication Server 1000 Release 4.5/Business Communications Manager

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