Enterprise: Common

Solution Integration Guide for Multisite Business Communications Manager Systems

NN49000-303

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Contents

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 Business Communications Manager (BCM), click one of the following links:

For the...	Go to...
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
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 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 configuration of the Business Communications Manager (BCM) to integrate multiple BCM systems in a network. Integrate the BCM systems when all systems are installed and a baseline of operation has been achieved and tested.

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

- Business Communications Manager 200 Releases 4.0, 3.7, and 2.0
- Business Communications Manager 400 Releases 4.0, 3.7, and 2.0
- 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 multisite 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.

If you are integrating the BCM to a CS 1000 system, refer to *Solution Integration Guide for Communication Server 1000/Business Communications Manager (NN43001-326)*.

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)
- *Keycode Installation Guide* (NN40010-301)

Overview

The tasks in the Business Communications Manager multisite integration process are listed in Table 1 "Task Completion Checklist" (page 9). Use this checklist to implement the integration.

Table 1
Task Completion Checklist

	Task	Reference
	Configure BCM 200/400 Release 4.0	1. "Configuring incoming VoIP trunks" (page 17) 2. "Verifying system license and keycodes" (page 18) 3. "Configuring VoIP trunk media parameters" (page 19) 4. "Configuring local Gateway parameters" (page 23) 5. "Configuring VoIP lines" (page 28) 6. "Configuring target lines" (page 33)
	Configure BCM 200/400 Release 3.7	1. "Verifying incoming VoIP trunks provisioning" (page 37) 2. "Adding keycodes files" (page 38) 3. "Adding a functionality-specific keycode" (page 38) 4. "Configuring VoIP H.323 trunk media parameters" (page 39) 5. "Configuring VoIP SIP trunk media parameters" (page 40) 6. "Configuring H.323 local Gateway IP parameters" (page 41) 7. "Configuring SIP local Gateway IP parameters" (page 42) 8. "Configuring SIP subdomains" (page 43) 9. "Configuring remote H.323 Gateways" (page 44) 10. "Configuring remote SIP endpoints" (page 45) 11. "Configuring VoIP lines for outgoing calls" (page 46)

	Task	Reference
		12. "Configuring target lines for incoming calls" (page 49) 13. "Configuring telephones to access outgoing VoIP lines" (page 50)
	Configure BCM50	1. "Configuring incoming VoIP trunks" (page 51) 2. "Verifying system license and keycodes" (page 52) 3. "Configuring VoIP trunk media parameters" (page 53) 4. "Configuring local Gateway parameters" (page 57) 5. "Configuring VoIP lines" (page 61) 6. "Configuring target lines" (page 65)

Prerequisites

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

- "Knowledge requirements" (page 11)
- "Capturing integration parameters" (page 11)
- "Establishing the system baseline" (page 12)

Knowledge requirements

The following knowledge and skills are required to implement a multisite BCM systems integration:

- 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

Training

Nortel recommends that you complete product-specific training before you begin integrating the BCM systems. A complete list of courses is available at www.nortel.com

Capturing integration parameters

Table 2 "Integration parameters" (page 11) 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	

Parameter	Value
SIP Gateway endpoint authentication password (must match the NRS password)	
IP addresses and URLs	
Gatekeeper IP address	
Alternate Gatekeeper IP address (optional)	
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)	
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	
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 Business Communications Manager (BCM) systems, so that the systems are configured and working in a stand-alone environment.

Use Table 3 "Pre-integration checklist" (page 13)to complete system baselines prior to integration.

Table 3
Pre-integration checklist

	Task	Reference	Comments
	BCM configuration is complete and passing data traffic.		
	BCM networking hardware is installed for integration.		To check the installed hardware: **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.
	PEC III Media Service Cards (MSC) are later.		PECIII MSCs are required for T.38 Fax and IP telephony. To check the PEC hardware: **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.

	Task	Reference	Comments
	BCM 200/400 is Release 4.0, 3.7, or 2.0. BCM50 is Release 2.0 or later.		To check the software version: **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.		To check the software version: **1** Log on to Element Manager. **2** Select the **Configuration** tab. **3** Expand the **System** folder. **4** Select **Identification**.
	VoIP Gateway Trunk licensing is purchased and loaded on BCM.	*Keycode Installation Guide* (NN40010-301)	To check Feature Licenses: **1** Log on to Element Manager. **2** Select the **Configuration** tab. **3** Expand the **System** folder. **4** Select **Keycodes**.
	IP Client licensing is purchased and loaded on BCM.	*Keycode Installation Guide* (NN40010-30	To check Feature Licenses: **1** Log on to Element Manager. **2** Select the **Configuration** tab. **3** Expand the **System** folder. **4** Select **Keycodes**.
	MCDN feature licensing is purchased and loaded on BCM.	*Keycode Installation Guide* (NN40010-30	To check Feature Licenses: **1** Log on to Element Manager.

	Task	Reference	Comments
			2 Select the **Configuration** tab. **3** Expand the **System** folder. **4** Select **Keycodes**.

BCM 200/400 Release 4.0 configuration

This chapter describes configuration procedures for the Business Communications Manager (BCM) 200 and 400 Release 4.0 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 Release 4.0 configuration procedures

The sequence of BCM 200/400 Release 4.0 configuration procedures is as follows:

- "Configuring incoming VoIP trunks" (page 17)
- "Verifying system license and keycodes" (page 18)
- "Configuring VoIP trunk media parameters" (page 19)
- "Configuring local Gateway parameters" (page 23)
- "Configuring VoIP lines" (page 28)
- "Configuring target lines" (page 33)

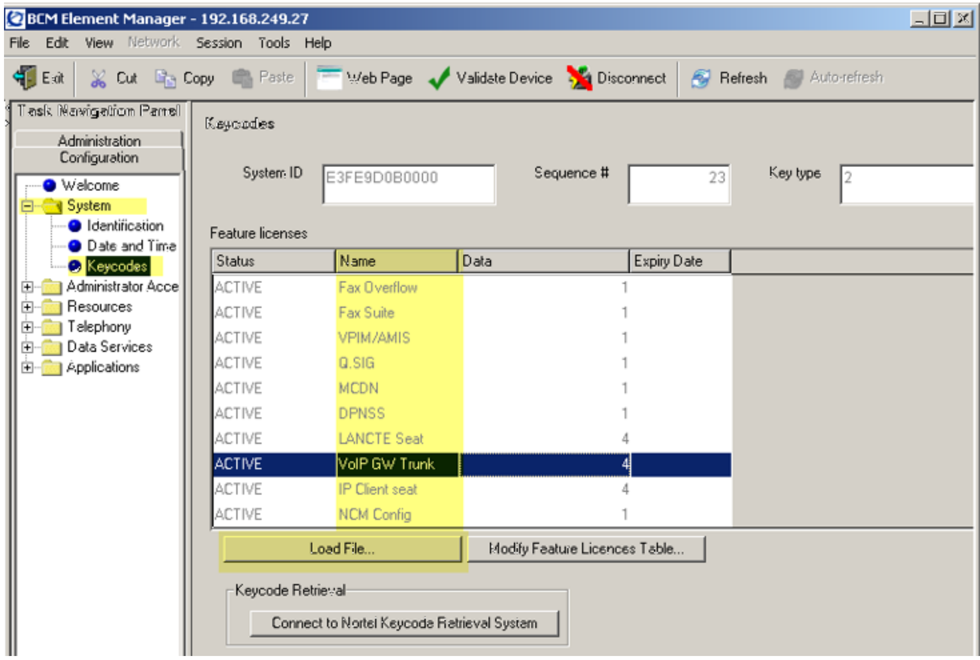
Configuring incoming VoIP trunks

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 Figure 1 "Keycodes" (page 18).

Figure 1
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 Figure 1 "Keycodes" (page 18).
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—

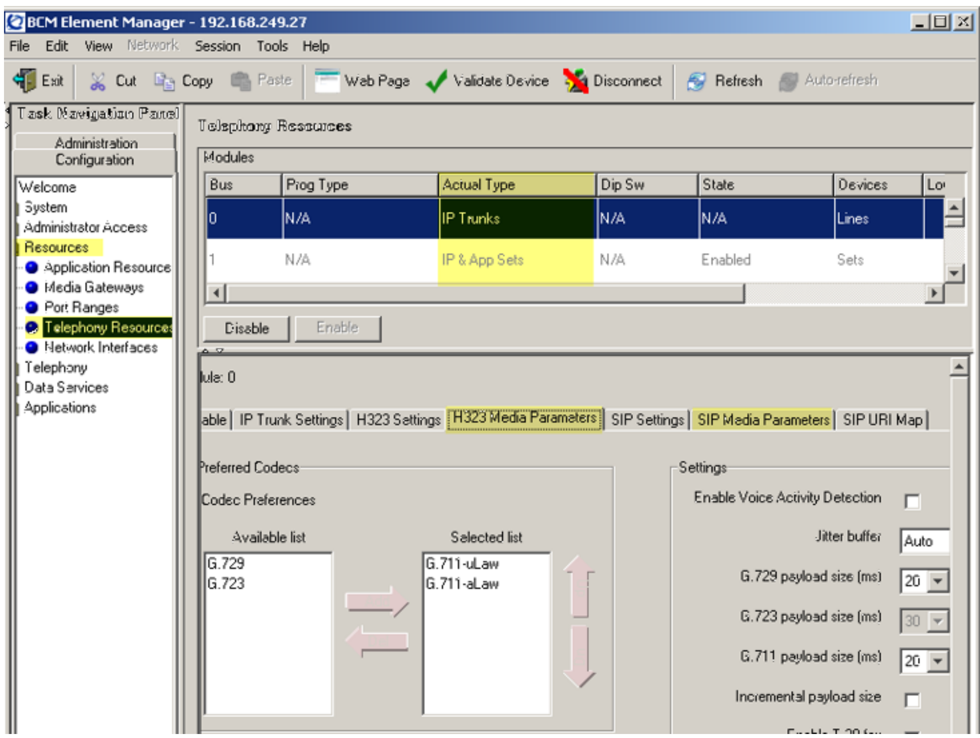
Configuring VoIP trunk media parameters

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 Figure 2 "Telephony Resources" (page 19).

Figure 2
Telephony Resources



Step	Action
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 Table 4 "H.323 Media Parameters fields" (page 20) and Table 5 "SIP Media Parameters fields" (page 21) for a description of the parameters.

—End—

Table 4
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 *BCM 4.0 Telephony Device Installation Guide* (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: • 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 5
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	**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.
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 *BCM 4.0 Telephony Device Installation Guide* (N0060609).
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

Field	Value	Description
		the fax machine. To minimize the possibility of your VoIP calls being dropped due to fax tone interference: • 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	Select **Resources > Telephony Resources**.
4	In the **Module Panel**, select the line in which the **Actual Type** column is set to **IP Trunks**. See Figure 2 "Telephony Resources" (page 19).
5	Select the **IP Trunk Settings** tab and enter the information that supports your system. See Figure 3 "IP Trunk Settings" (page 24). Refer to Table 6 "IP Trunk Settings fields" (page 24) for information about the IP Trunk Settings fields.

Figure 3
IP Trunk Settings

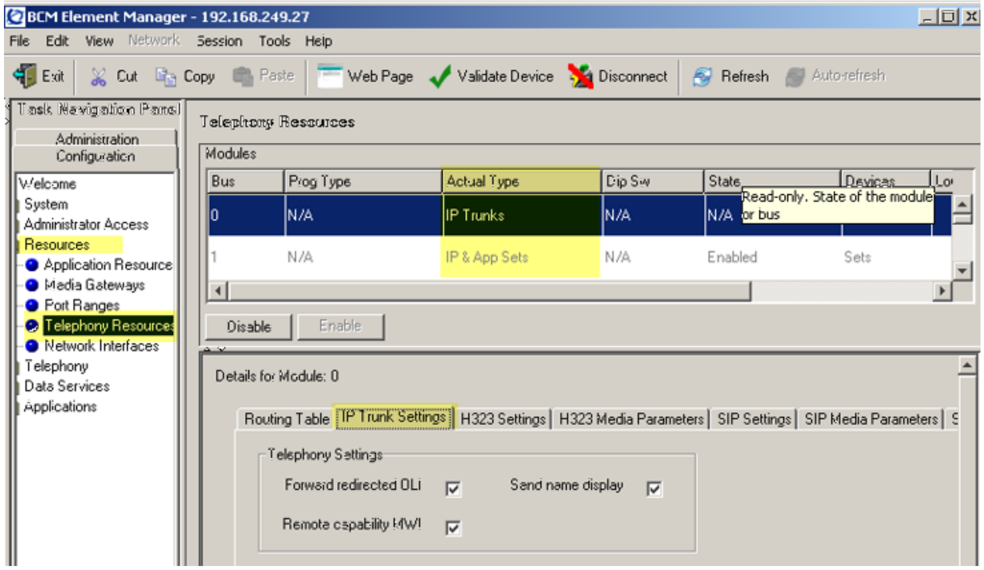
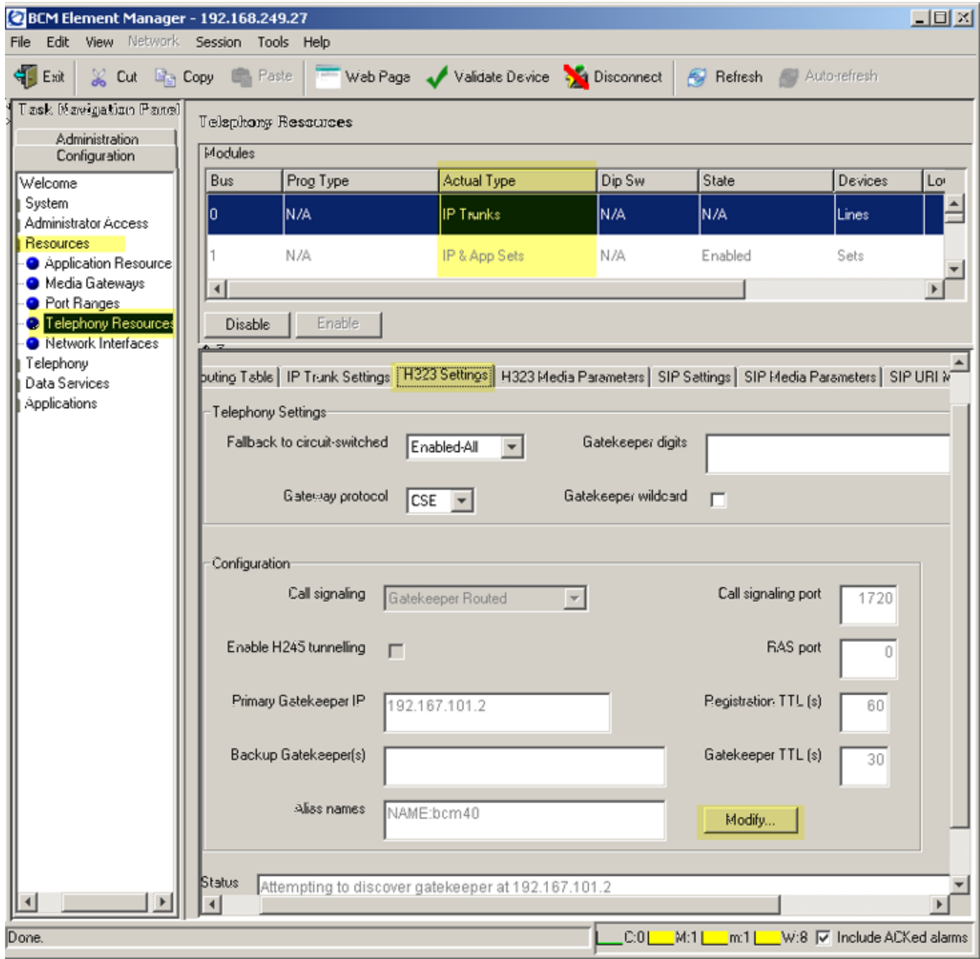


Table 6
IP Trunk Settings fields

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.

6 For H.323 VoIP trunks, select the **H.323 Settings** tab. See Figure 4 "H.323 Settings" (page 25).

Figure 4
H.323 Settings



7 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.

8 For **Gateway protocol**, select **CSE**.

9 Scroll down to **Alias names** and click **Modify**.
The Modify Call Signaling Settings page appears.

10 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 Table 7 "H.323 Call Signaling Settings fields" (page 26).

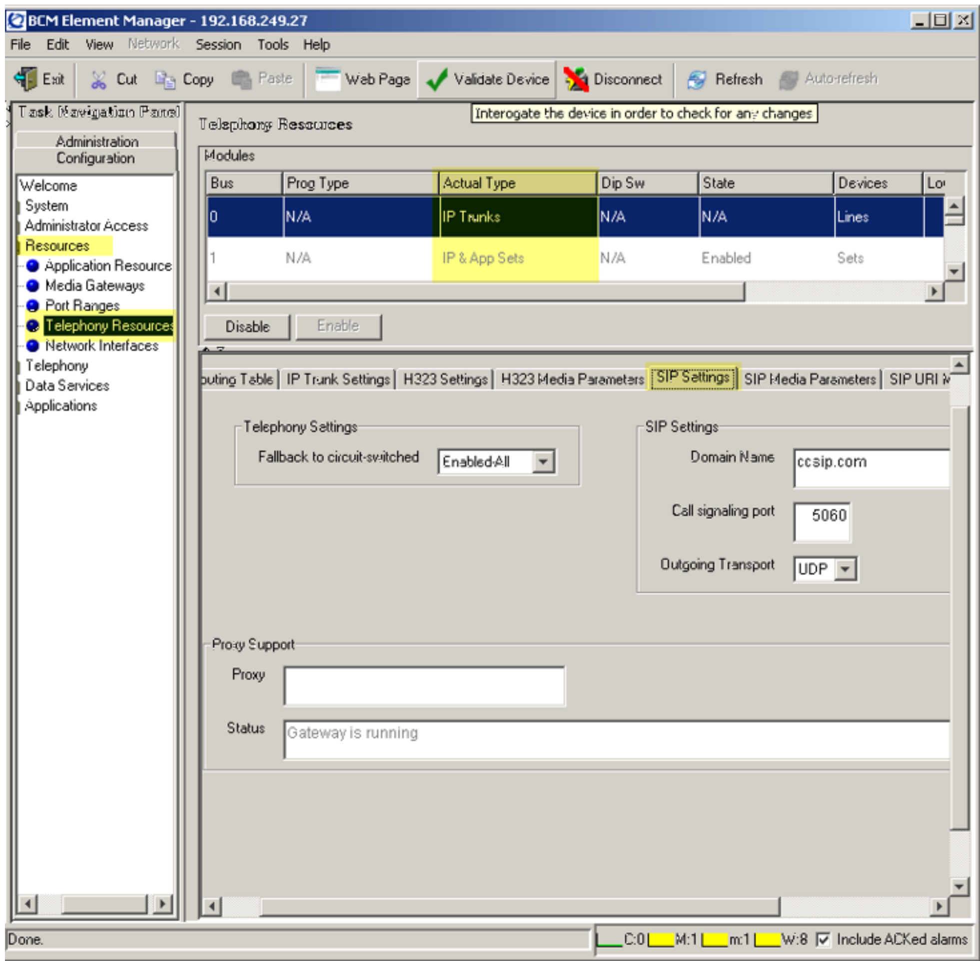
Table 7
H.323 Call Signaling Settings fields

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.

11 For SIP trunks, select the **SIP Settings** tab. See Figure 5 "SIP Settings" (page 27).

Figure 5
SIP Settings



12 Enter the information that supports your system.

Refer to Table 8 "SIP Settings fields" (page 28) for more information.

Table 8
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-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.
	Enabled-All	
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—

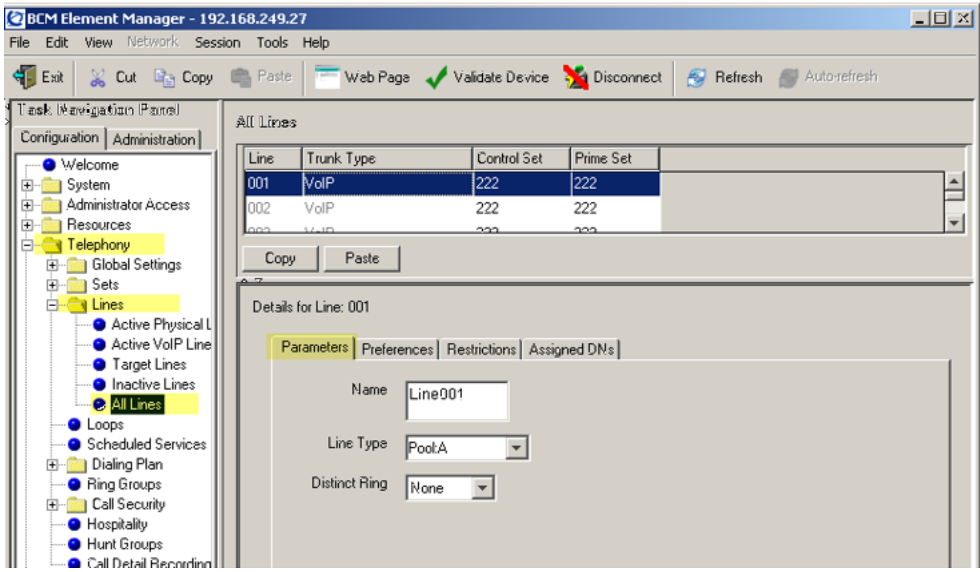
Configuring VoIP lines

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 Figure 6 "VoIP lines" (page 29).

Figure 6
VoIP lines



6 Configure the Parameters tab appropriately for your network. Refer to Table 9 "VoIP line descriptions" (page 29) for configuration information.

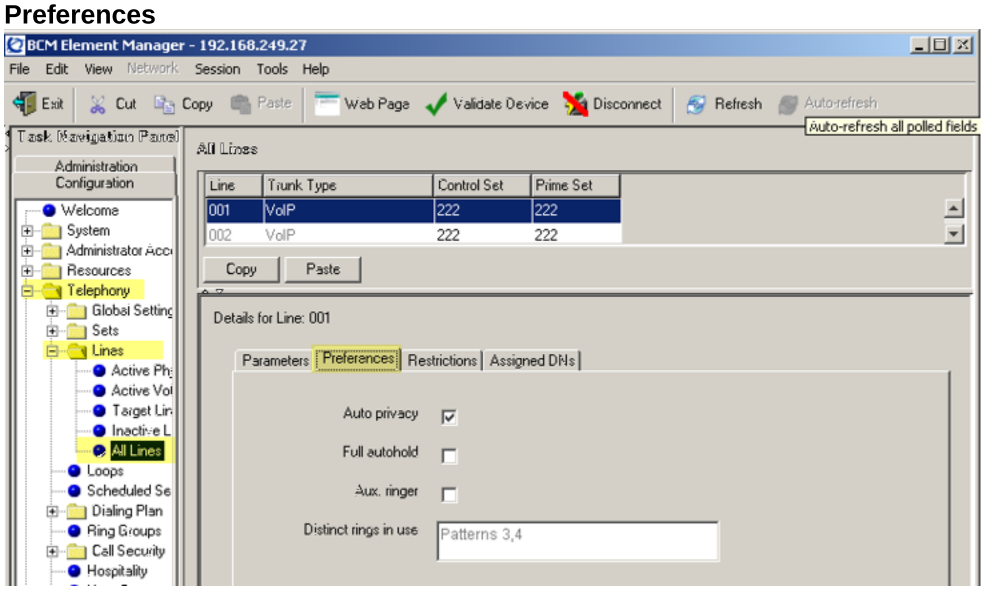
Table 9
VoIP line descriptions

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		Defines how the line is used in relation to other lines in the system.
	Public	If the line is to be shared among telephones, set to Public.
	DN:*	If the line is assigned to only one telephone, set to DN:*.
	Pool [A to O]	If you are using routing, put the line into line pool (A to F). 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.
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 Figure 7 "Preferences" (page 30).

Figure 7
Preferences



8 Configure the Preferences tab appropriately for your network.

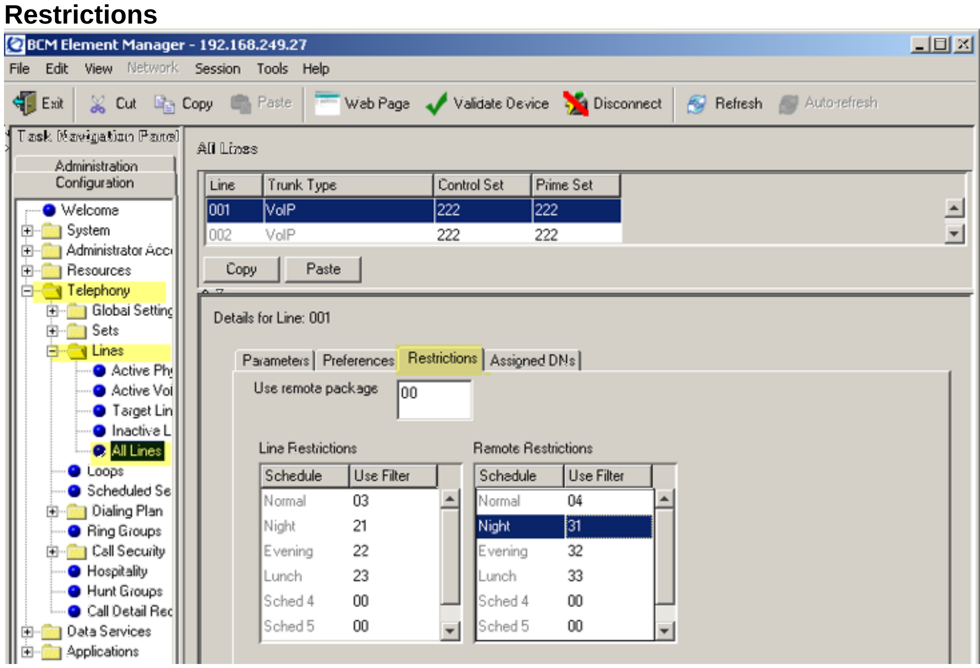
Refer to Table 10 "Preferences fields" (page 31) for configuration information.

Table 10
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 *BCM 4.0 Device Configuration Guide* (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 Figure 8 "Restrictions" (page 31).

Figure 8
Restrictions



10 Configure the Restrictions tab appropriately for your network.

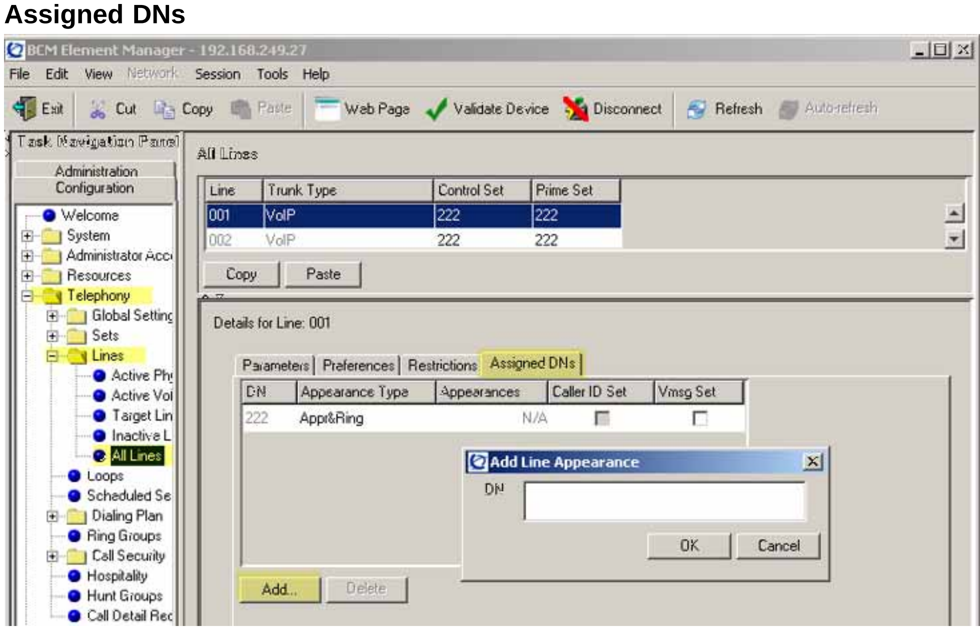
Refer to Table 11 "Restrictions fields" (page 32) for configuration information.

Table 11
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 Restriction s - 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 Figure 9 "Assigned DNs" (page 32).

Figure 9
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 Table 12 "Assigned DNs fields" (page 33) for configuration information.

Table 12
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.

5 Select the **Parameters** tab and enter the appropriate information for your network.
See Figure 10 "Parameters" (page 34). Refer to Table 13 "Parameters fields" (page 34) for configuration information.

Figure 10
Parameters

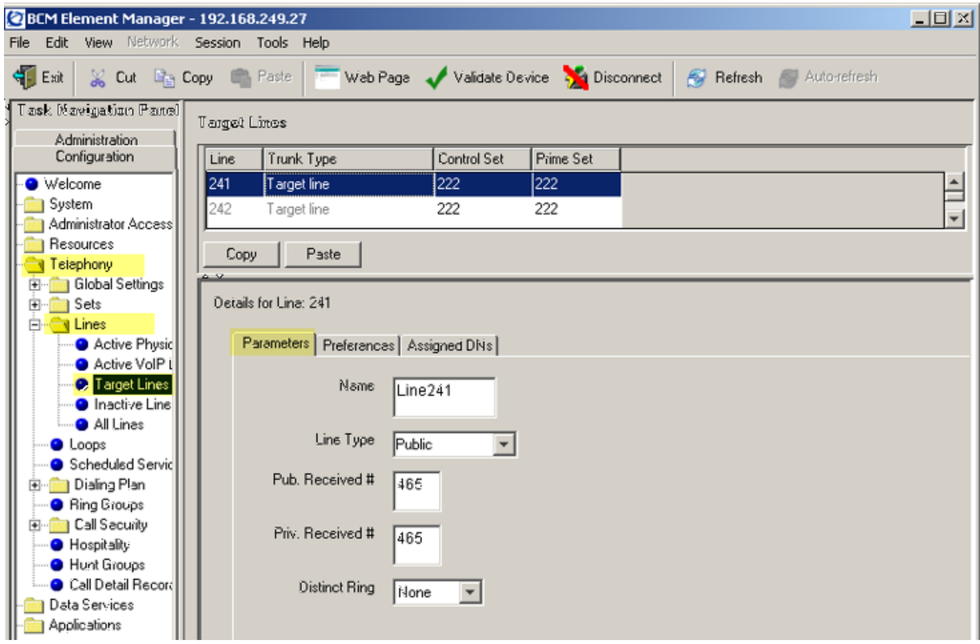


Table 13
Parameters fields

Field	Value	Description
Name		Enter the name for the line, for example, Line241.
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 Figure 11 "Preferences" (page 35). Refer toTable 14 "Preferences fields" (page 35) for configuration information.

Figure 11
Preferences

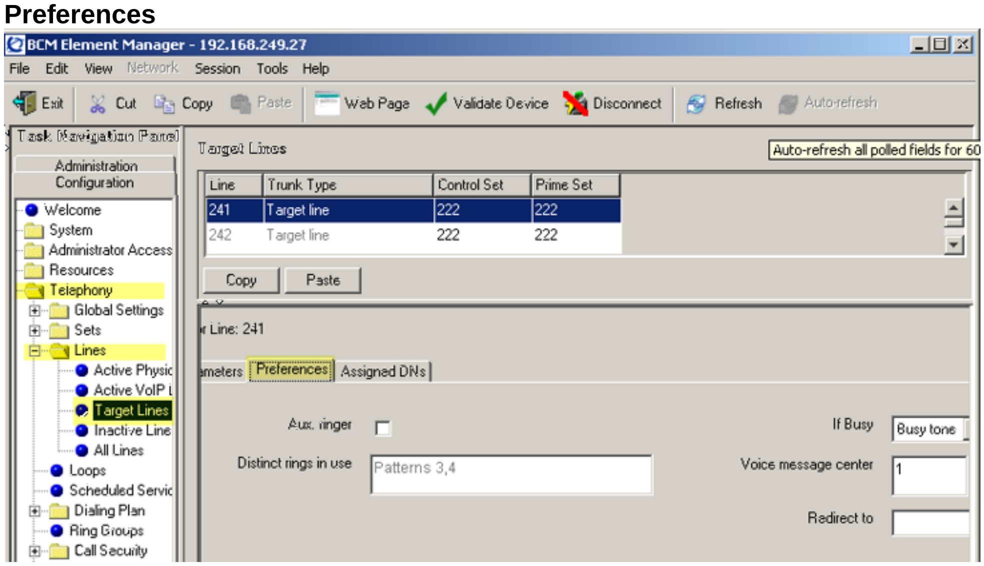


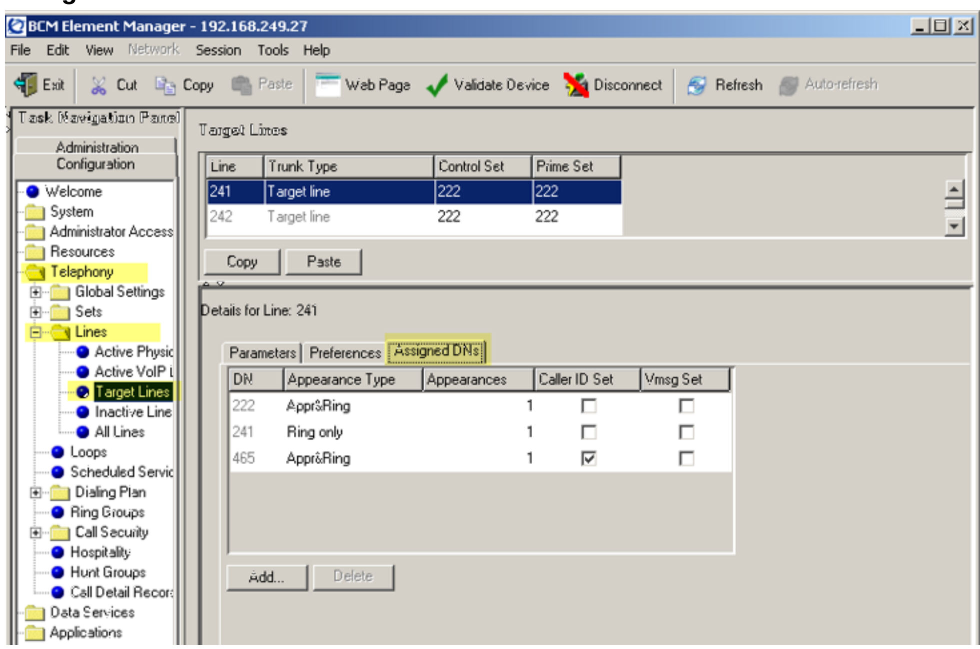
Table 14
Preferences fields

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 Figure 12 "Assigned DNs" (page 36).

Figure 12
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 Table 12 "Assigned DNs fields" (page 33) for configuration information.

—End—

BCM 200/400 Release 3.7 configuration

This chapter describes configuration procedures for the Business Communications Manager (BCM) 200 and 400 Release 3.7 systems.

BCM 200/400 Release 3.7 configuration procedures

The sequence of BCM 200/400 Release 3.7 configuration procedures is as follows:

Verifying incoming VoIP trunks provisioning

Perform this procedure to verify that incoming VoIP trunks are provisioned.

Step	Action
1	Log on to the Unified Manager.
2	Select the **BCM>System>Licensing** heading. The Licensing Setting page appears.

Step	Action
3	Select the Applied Keycodes tab.
4	In the list of applied keycodes, check that there are sufficient VoIP gateway ports.

—End—

Adding keycodes files

Perform the following procedure to add keycodes.

Step	Action
1	Log on to the Unified Manager.
2	Select **BCM>System>Licensing>Keycode Files**. The Keycode File Location Information page appears.
3	Enter the required information for the keycode file.
4	Select the **Configuration** tab.
5	Click **Apply new Keycode File**.
6	A message appears asking you to confirm. Click **Yes**.
7	When prompted, reboot the system to activate your new keycodes.

—End—

Adding a functionality-specific keycode

Perform the following procedure to verify the system license and keycodes.

Step	Action
1	Log on to the Unified Manager.
2	Select the **BCM>System>Licensing** heading. See figure from Adding keycodes files.
3	Select the **Configuration** tab.
4	Click **Add a keycode**. The Keycode dialog box appears.
5	Enter a valid **Keycode**.
6	Click **Save**.

—End—

Configuring VoIP H.323 trunk media parameters

Perform the following procedure to configure H.323 Gateway trunks.

Step	Action
1	Log on to the Unified Manager.
2	Select the **BCM>Services>IP telephony>IP trunks>H.323 trunks** heading. The Local Gateway IP Interface page appears.
3	Select the **Media Parameters** tab.
4	Configure the parameters listed in the table below with the appropriate values for your network. Ensure that these settings are consistent with the other systems on your network

Table 15
H.323 media parameters

Parameter	Value
1st Preferred Codec	None G.729
2nd Preferred Codec	None G.723
3rd Preferred Codec	None G.711-uLaw
4th Preferred Codec	None G.711-aLaw
Silence Compression	Enabled Disabled
Jitter Buffer – Voice	Auto None Small Medium Large

Parameter	Value
T.38 Fax Support	Enabled Disabled
G.729 Payload Size (ms)	10, 20, 30, 40, 50, 60
G.723 Payload Size (ms)	30
G.729 Payload Size (ms)	10, 20, 30, 40, 50, 60
Incremental Payload Size	Enabled Disabled

—End—

Configuring VoIP SIP trunk media parameters

Perform the following procedure to configure SIP media parameters.

Step	Action
1	Log on to the Unified Manager.
2	Select the **BCM>Services>IP Telephony> IP Trunks>SIP Trunks** heading. The SIP Trunks Summary page appears.
3	Select the **Media Parameters** tab.
4	Configure the parameters listed in the table below with the appropriate values for your network. Ensure that these settings are consistent with the other systems on your network.

Table 16
SIP media parameters

Parameter	Value
1st Preferred Codec	None G.729
2nd Preferred Codec	None G.723
3rd Preferred Codec	None G.711-uLaw

Parameter	Value
4th Preferred Codec	None G.711-aLaw
Silence Compression	Enabled Disabled
Jitter Buffer – Voice	Auto None Small Medium Large

—End—

Configuring H.323 local Gateway IP parameters

Perform the following procedure to configure local Gateway parameters.

Step	Action
1	Log on to the Unified Manager.
2	Select the **BCM>Services>IP Telephony>IP Trunks>H.323 Trunks** heading. The Local Gateway IP Interface page appears.
3	Select **Resources>Telephony Resources**.
4	In the Local Gateway IP Interface section, configure the parameters listed in the table below with the appropriate values for your network.

Table 17
H.323 local Gateway IP parameters

Parameter	Value
Fallback to Circuit-Switched	Enabled-All Enabled-TDM-only Disabled

Parameter	Value
Call Signaling	Direct GatekeeperRouted GatekeeperResolved Gatekeeper RoutedNoRAS
Primary Gatekeeper IP	
Backup Gatekeeper	
Alias Names	
Registration TTL (Seconds)	
Gateway Protocol	None SL1 CSE
H245 Tunneling	Enabled Disabled
Call Signaling Port	
RAS Port	
Force G.711 for 3.1k Audio	Enabled Disabled
Forward Redirected OLI	Enabled Disabled

5 When implementing your dialing plan, in the **H.323 Local Gateway IP Interface** tab, be sure to select a value for **Fall back to circuit-switched**. This determines how the system handles calls if the IP network cannot be used.

6 For **Gateway protocol**, select **CSE**.

7 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.

—End—

Configuring SIP local Gateway IP parameters

Perform the following procedure to configure SIP local Gateway IP parameters.

Step	Action
1	Log on to the Unified Manager.
2	Select the **BCM>Services>IP Telephony>IP Trunks>SIP Trunks** heading. The Summary page appears.
3	Configure the parameters listed in the table below with the appropriate values for your network.

Table 18
SIP local Gateway IP parameters

Parameter	Value
Fallback to Circuit-Switched	Enabled-All Enabled-TDM-only Disabled
SIP Domain	

—End—

Configuring SIP subdomains

Perform the following procedure to configure SIP subdomains.

Step	Action
1	Log on to the Unified Manager.
2	Expand the **BCM>Services>IP telephony>IP trunks>SIP trunks** heading. The SIP Trunks Summary page appears.
3	Select the **Dialing Sub-Domain** tab.
4	Configure the parameters listed in the table below with the appropriate values for your network.

Parameter	Value
e.164 / National	
e.164 / Subscriber	
e.164 / Special	
e.164 / Unknown	

Parameter	Value
Private / UDP	
Private / CDP	
Private / Special	
Private / Unknown	
Unknown / Unknown	

—End—

Configuring remote H.323 Gateways

Perform the following procedure to configure remote H.323 Gateways.

Step	Action
1	Log on to the Unified Manager.
2	Expand the **BCM>Services>IP telephony>IP trunks>H.323 trunks>Remote Gateway** heading. The Remote Gateway page appears.
3	Select **Configuration**.
4	Select **Add Entry** to add a new remote gateway.
5	Configure the parameters listed in the table below with the appropriate values for your network.

Parameter	Value
Name	<alphanumeric>
Destination IP	<IP address>
QoS Monitor	Disabled Enabled
Transmit Threshold	0.0 (bad) to 5.0 (excellent)
Receive Threshold	0.0 (bad) to 5.0 (excellent)

Parameter	Value
Gateway type	BCM3.6 BCM3.5 BCM3.0 BCM2.5 CS1000 CS2000 IPT NetMeeting Norstar IP Gateway Other
Gateway Protocol	None SL1 CSE
Destination Digits	<numeric> Can be the same as the destination code for the route to the system.

6 Click the **Save** button to save the remote gateway.

—End—

Configuring remote SIP endpoints

Perform the following procedure to configure remote SIP endpoints.

Step	Action
1	Log on to the Unified Manager.
2	Expand the **BCM>Services>IP telephony>IP trunks>SIP trunks** heading.
3	Select **Address Book**. The Address Book page appears.
4	Select **Configuration**.
5	Select **Add Entry** to add a new remote gateway.

6 Configure the parameters listed in the table below with the appropriate values for your network.

Parameter	Value
Name	<alphanumeric>
Destination IP	<IP address>
QoS Monitor	Disabled Enabled
Transmit Threshold	0.0 (bad) to 5.0 (excellent)
Receive Threshold	0.0 (bad) to 5.0 (excellent)
Destination Digits	<numeric> Can be the same as the destination code for the route to the system.

7 Click the **Save** button to save the remote endpoint.

—End—

Configuring VoIP lines for outgoing calls

Perform the following procedure to configure VoIP lines for outgoing calls.

Step	Action
1	Log on to the Unified Manager.
2	Expand the **BCM>Services>Telephony Services>Lines>VoIP Lines>All VoIP lines** heading.
3	In the All VoIP lines section, expand the **Line** you wish to configure (for example, Line 001).
4	Select the **General** tab.
5	Configure the parameters listed in the table below with the appropriate values for your network.

Table 19
General parameters

Parameter	Value
Name	
Control Set	
Use Remote Package	

6 Expand the **Trunk/line data** heading.
The Trunk/Line data page appears.

7 Configure the parameters listed in the table below with the appropriate values for your network.

Note: The Line pool must belong to a line pool that contains the same type of VoIP line.

If you want specific restrictions assigned to the line, enter the information under the **Restrictions** heading.

Parameter	**Value**	**Description**
Line Type	Public Private to: Pool {A to O }	
Prime Set	DN: None DN <defined DN #>	
Distinct Ring	None Pattern 2 Pattern 3 Pattern 4	
Auto Privacy	N (No) Y (Yes)	
Use auxiliary ringer	N (No) Y (Yes)	
Full autohold	N (No) Y (Yes)	
Redirect to	<dial string>	Enter a dial string (including routing code) to redirect the line to an external telephone, such as a call attendant on another system. To stop redirection, delete the dial string and allow the record to update

8 Expand the **Restrictions>Line Restrictions** heading.

9 Configure the local restrictions schedules for this line. Refer to the table below for details.

Schedule	Filter	Description
Normal		Assign the filter to be used for Normal.
Night		Assign the filter to be used for Night.
Evening		Assign the filter to be used for Evening.
Lunch		Assign the filter to be used for Lunch.
Sched 4		Assign the filter to be used for Sched 4.
Sched 5		Assign the filter to be used for Sched 5.
Sched 6		Assign the filter to be used for Sched 6.

10 Expand the **Restrictions> Remote Restrictions** heading.

11 Configure the remote restrictions schedules for this line. Refer to the table below for details.

Schedule	Filter	Description
Normal		Assign the filter to be used for Normal.
Night		Assign the filter to be used for Night.
Evening		Assign the filter to be used for Evening.
Lunch		Assign the filter to be used for Lunch.
Sched 4		Assign the filter to be used for Sched 4.
Sched 5		Assign the filter to be used for Sched 5.
Sched 6		Assign the filter to be used for Sched 6.

12 Repeat this procedure for all the outgoing lines you wish to configure.

Note: Configuring SIP and H.323 trunks in the same pool may result in unpredictable results because they do not support the same level of service.

—End—

Configuring target lines for incoming calls

Perform the following procedure to configure telephones to access outgoing VoIP lines.

Step	Action
1	Log on to the Unified Manager.
2	Expand the **BCM>Services>Lines>Target Lines** heading.
3	Expand the target line to be configured.
4	Select the **General** tab.
5	Configure the parameters listed in the table below with the appropriate values for your network.

Table 20
Target line parameters

Parameter	Value
Name	
Control Set	

6	Select the **Trunk/Line data** tab.
7	Configure the parameters listed in the table below with the appropriate values for your network.

Table 21
Target line parameters

Parameter	Value
Trunk type	
Line Type	
If busy	
Prime Set	
Distinct ring in use	

Parameter	Value
Distinct ring	
use Auxiliary ringer	
redirect to	

—End—

Configuring telephones to access outgoing VoIP lines

Perform the following procedure to configure telephones to access outgoing VoIP lines.

Step	Action
1	Log on to the Unified Manager.
2	Expand the **BCM** folder.
3	Expand the **Services** heading.
4	Expand the **Telephony Services** heading.
5	Expand the **System DNs** heading.
6	Expand the **All System DNs** or **Active Set DNs** heading.
7	Expand the **DN** you wish to configure to use VoIP trunking (for example, DN 222).
8	Expand the **Line Access** heading.
9	Select **Line pool access**.
10	Click the **Add** button. The Add Line Pool Access page appears.
11	Type the letter of the VoIP Line Pool to be used.
12	Click **Save**.
13	To configure the line to access both H.323 and SIP Line pools, repeat steps 11 and 12.
14	Repeat this procedure for all telephones you wish to configure to access outside lines.

—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" (page 51)
- "Verifying system license and keycodes" (page 52)
- "Configuring VoIP trunk media parameters" (page 53)
- "Configuring local Gateway parameters" (page 57)
- "Configuring VoIP lines" (page 61)
- "Configuring target lines" (page 65)

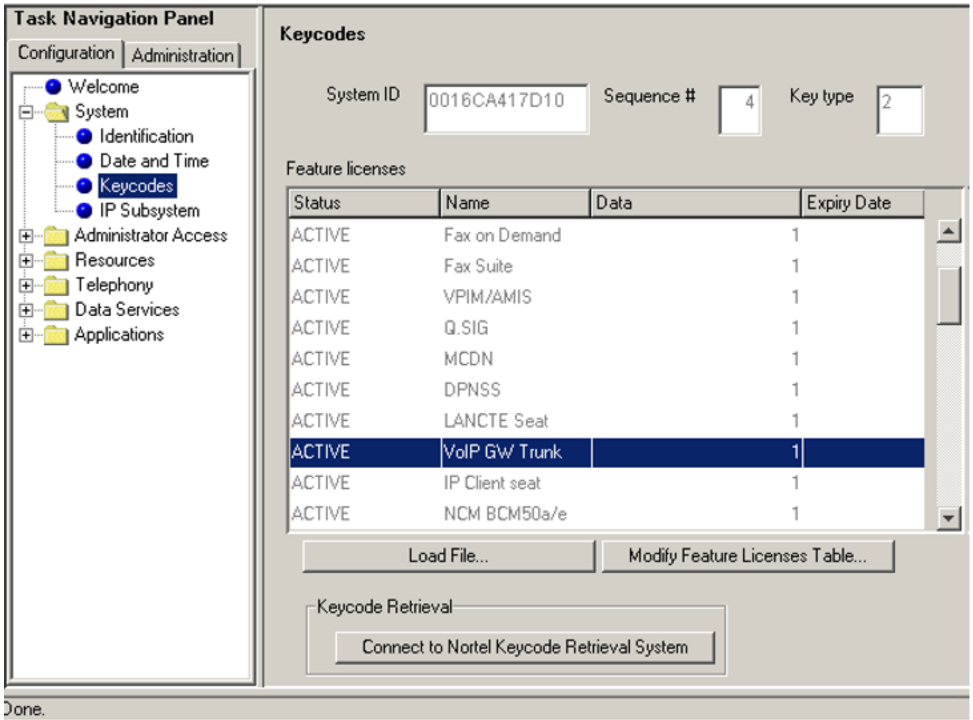
Configuring incoming VoIP trunks

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 Figure 13 "Keycodes" (page 52).

Figure 13
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 Figure 13 "Keycodes" (page 52).

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—

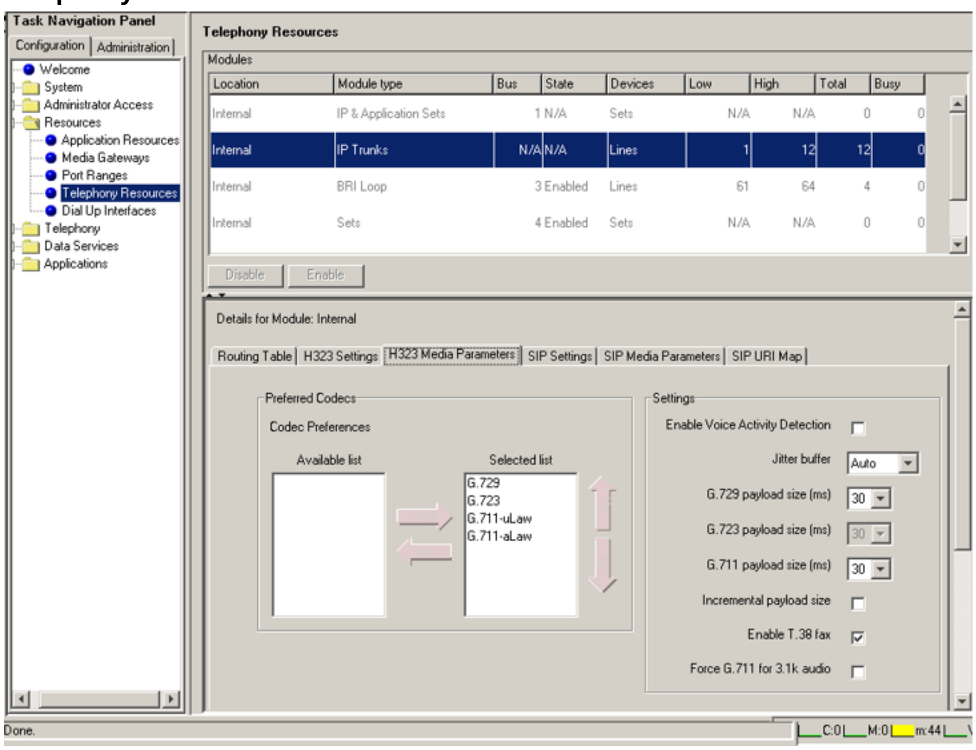
Configuring VoIP trunk media parameters

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 Figure 14 "Telephony Resources" (page 53).

Figure 14
Telephony Resources



Step	Action
4	In the **Modules** panel, select the line where the **Module 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 Table 22 "H.323 Media Parameters fields" (page 54) and Table 23 "SIP Media Parameters fields" (page 55) for a description of the parameters.

—End—

Table 22
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 *BCM 4.0 Telephony Device Installation Guide* (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: • 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 23
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	**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.
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 *BCM 4.0 Telephony Device Installation Guide* (N0060609).
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

Field	Value	Description
		the fax machine. To minimize the possibility of your VoIP calls being dropped due to fax tone interference: • place the fax machine away from other telephones • turn the fax machine's speaker volume to the lowest level, or off, if available

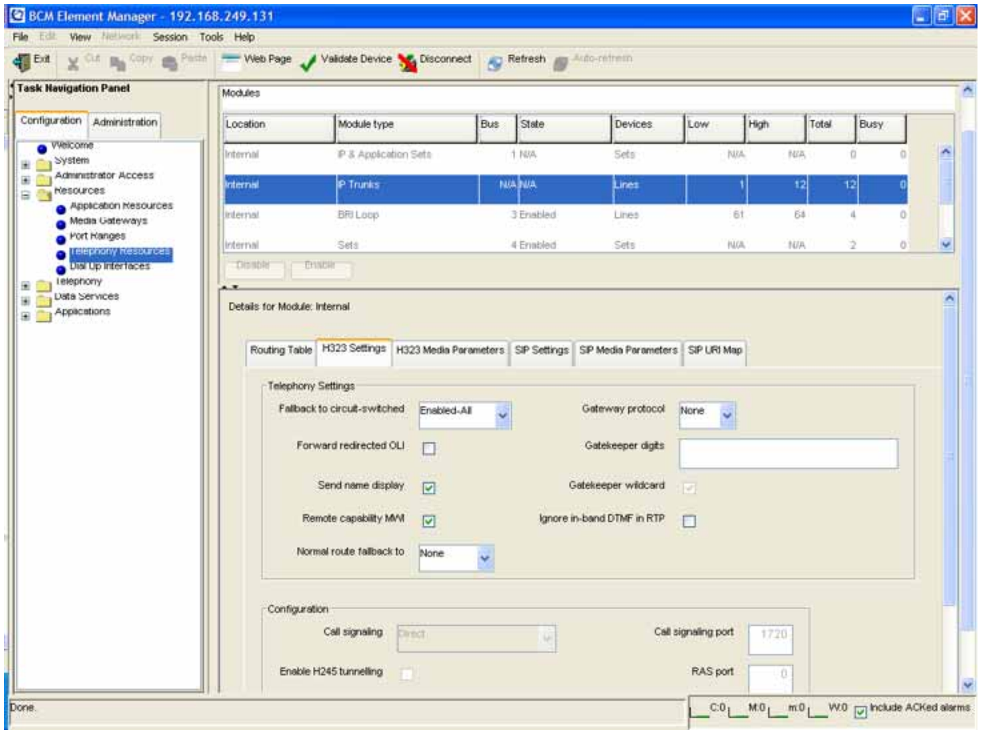
Configuring local Gateway parameters

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	Select **Resources > Telephony Resources**.
4	In the **Module Panel**, select the line in which the **Module type** column is set to **IP Trunks**. See Figure 14 "Telephony Resources" (page 53).
5	For H.323 VoIP trunks, select the **H.323 Settings** tab. See Figure 15 "H323 Settings" (page 58).

Figure 15
H323 Settings



6 When implementing your dialing plan, in the **H323 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 names** 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 Table 24 "H.323 Call Signaling Settings fields" (page 59).

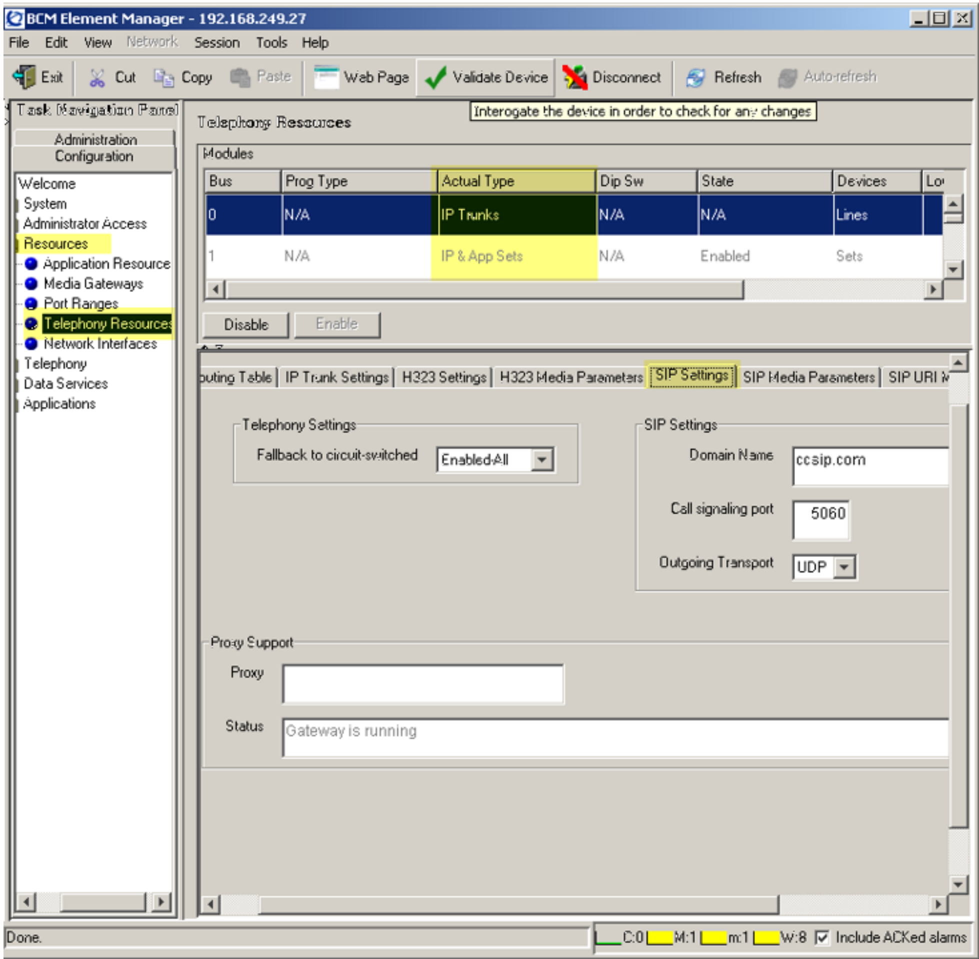
Table 24
H.323 Call Signaling Settings fields

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 Gatekeepe r(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 Figure 16 "SIP Settings" (page 60).

Figure 16
SIP Settings



11 Enter the information that supports your system.

Refer to Table 25 "SIP Settings fields" (page 61) for more information.

Table 25
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-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.
	Enabled-All	
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—

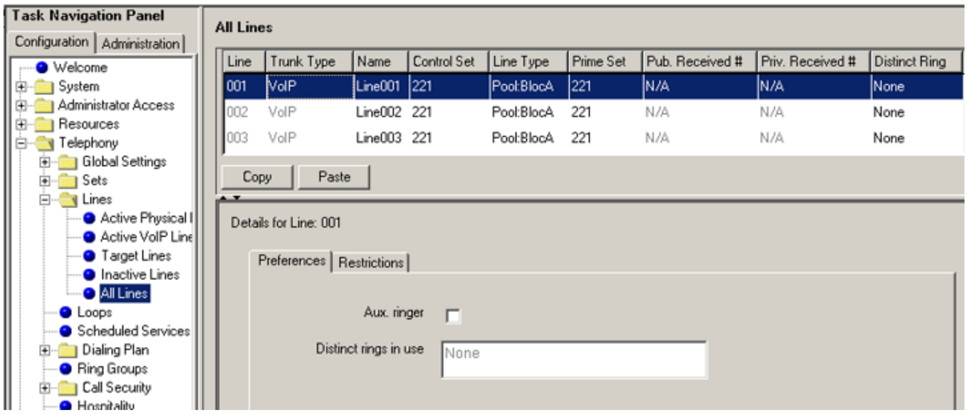
Configuring VoIP lines

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 Figure 17 "Preferences" (page 62).

Figure 17
Preferences



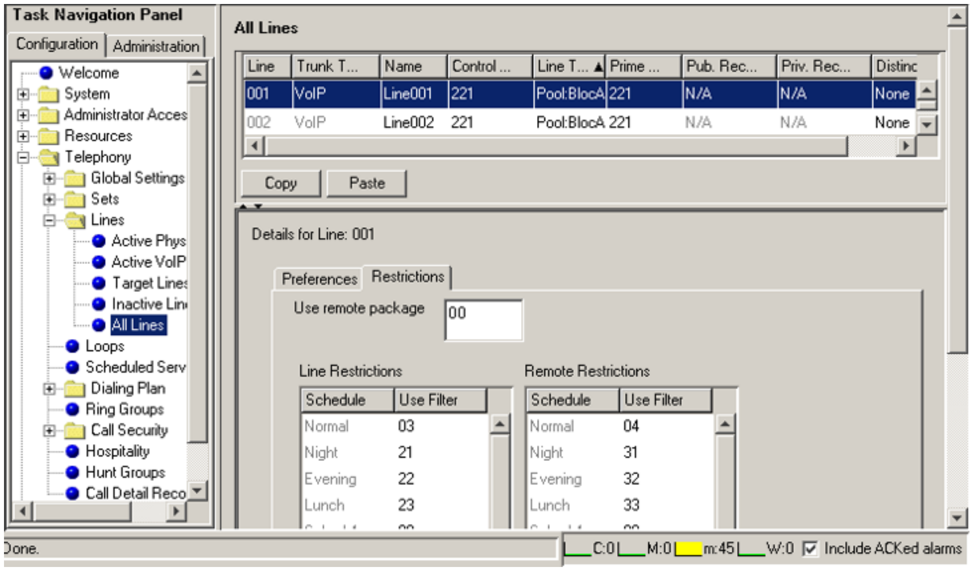
6 Configure the Preferences tab appropriately for your network. Refer to Table 26 "Preferences fields" (page 62) for configuration information.

Table 26
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 Figure 18 "Restrictions" (page 63).

Figure 18
Restrictions



8 Configure the Restrictions tab appropriately for your network. Refer to Table 27 "Restrictions fields" (page 63) for configuration information.

Table 27
Restrictions fields

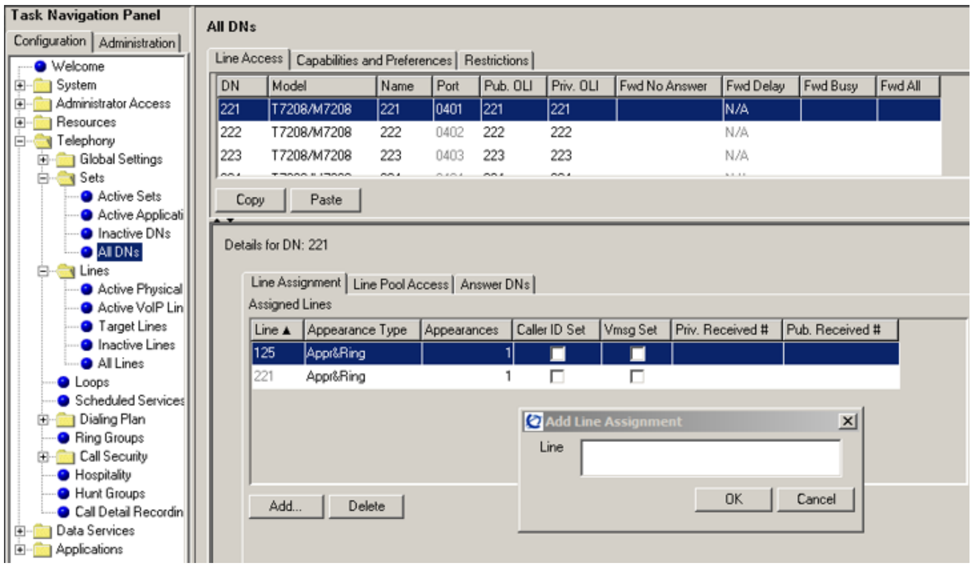
Field	Value	Description
Use remote package	< package #>	If the line is used to receive external calls or calls from other nodes on the private network, ensure that you indicate a remote package that provides only the availability that you want for external callers. This attribute is typically used for 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 > Sets > All DNs**.

10 Highlight the individual line you wish to configure.

11 Select the **Line Assignment** tab.
See Figure 19 "Line Assignment" (page 64).

Figure 19
Line Assignment



12 Edit the listed DNs, or click **Add** to add a DN as required.

13 Enter the appropriate information for your network.
Refer to Table 28 "Assigned DNs fields" (page 64) for configuration information.

Table 28
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.

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 Figure 20 "Preferences" (page 66). Refer toTable 29 "Preferences fields" (page 66) for configuration information.

Figure 20
Preferences

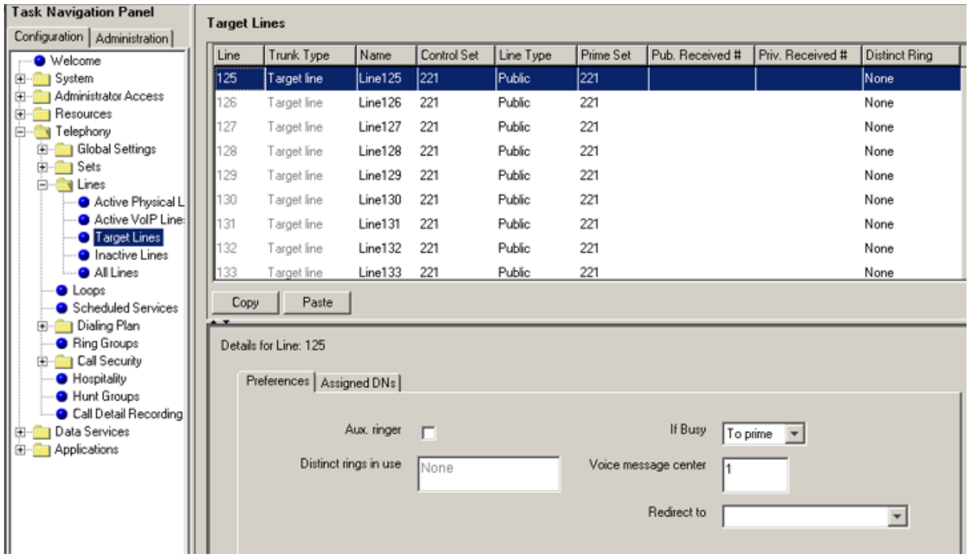
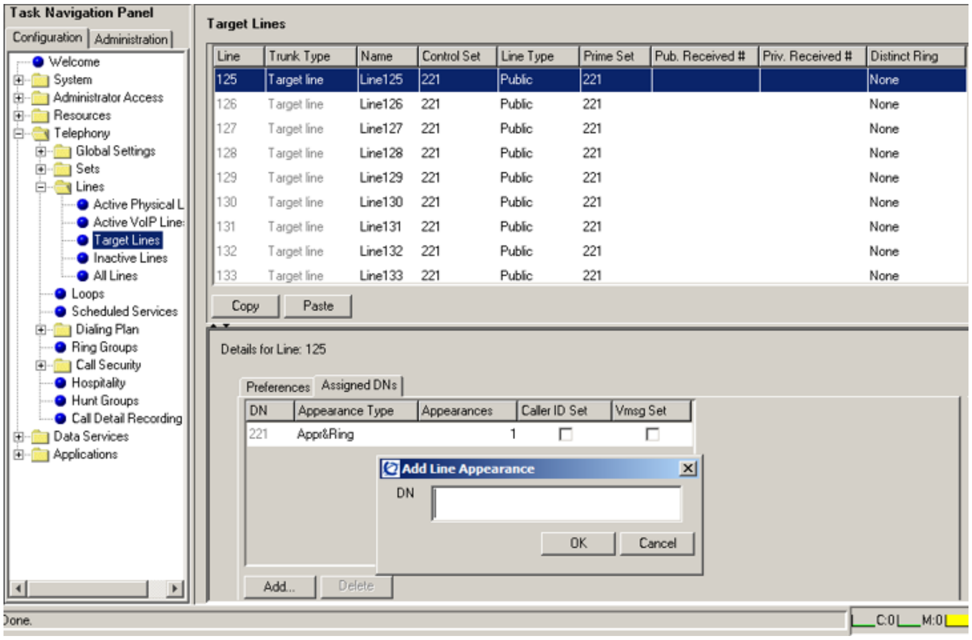


Table 29
Preferences fields

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 DNs** tab.
See Figure 21 "Assigned DNs" (page 67).

Figure 21
Assigned DNs



7 Edit the listed DNs, or click **Add** to add a DN as required.

8 Enter the appropriate information for your network.
Refer to Table 12 "Assigned DNs fields" (page 33) for configuration information.

—End—

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