CHAPTER 71

Trunk Configuration

Use a trunk device to configure a logical route to a gatekeeper (that is, the wholesale network or an intercluster trunk with gatekeeper control), to an intercluster trunk without a gatekeeper, or to a SIP network. Choose from the following available trunk types:

- H.225 trunk (gatekeeper controlled)
- Intercluster trunk (gatekeeper controlled)
- Intercluster trunk (non-gatekeeper controlled)
- SIP trunk

The following topics cover Cisco Unified CallManager trunk configuration:

- Finding a Trunk, page 71-1
- Configuring a Trunk, page 71-2
- Trunk Configuration Settings, page 71-3
- Deleting a Trunk, page 71-26
- Resetting a Trunk, page 71-27

The following topics contain additional information that is related to trunks:

- Call Admission Control, *Cisco Unified CallManager System Guide*
- Gatekeepers and Trunks, *Cisco Unified CallManager System Guide*
- Gatekeeper and Trunk Configuration in Cisco Unified CallManager, *Cisco Unified CallManager System Guide*
- *Cisco Unified Communications Solution Reference Network Design (SRND)*

Finding a Trunk

Because you might have multiple trunks in your network, Cisco Unified CallManager lets you search for trunks on the basis of specified criteria. Follow these steps to search for a specific trunk in the Cisco Unified CallManager database.

Note During your work in a browser session, Cisco Unified CallManager Administration retains your trunk search preferences. If you navigate to other menu items and return to this menu item, Cisco Unified CallManager Administration retains your trunk search preferences until you modify your search or close the browser.

Procedure

Step 1 Choose **Device > Trunk**.

The Find and List Trunks window displays.

Step 2 Choose the field that you want to use to locate a trunk.

Note To find all trunks that are registered in the database, choose Device Name from the list of fields and choose "is not empty" from the list of patterns; then, click **Find**.

Step 3 Choose the appropriate search pattern for your text search. If you do not want to perform a text search, choose "is empty."

Step 4 Enter your search text, if any, in the Find field.

Step 5 If you choose calling search space or device pool in Step 2, the options available in the database display. From the drop-down list box below the **Find** button, you can choose one of these options.

Step 6 Click **Find**.

A list of devices that match the criteria displays. This window also lists the total number of devices.

Step 7 To view the next set of discovered devices, click **Next**.

Note You can delete or reset multiple trunks from the Find and List Trunks window by checking the check boxes next to the appropriate trunks and clicking **Delete Selected** to delete the trunks or clicking **Reset Selected** to reset the trunks. You can choose all of the trunks in the window by clicking **Select All**.

Additional Topics

See the "Related Topics" section on page 71-28.

Configuring a Trunk

Perform the following procedure to add a new trunk device or update an existing trunk device.

Note You can configure multiple trunk devices per Cisco Unified CallManager cluster.

Before You Begin

Configure SIP Trunk Security Profiles and SIP Profiles before you configure a SIP Trunk. For more information, see the "Configuring SIP Profiles" section on page 79-2, SIP Trunk Security Profile Configuration, and the *Cisco Unified CallManager Security Guide*.

Procedure

Step 1 Choose **Device** > **Trunk**.

The Find and List Trunks window displays.

Step 2 Perform one of the followings tasks:

- To add a new trunk device, click the **Add New** button. The Trunk Configuration window displays. Continue with Step 3.
- To update trunk settings, locate the appropriate trunk as described in "Finding a Trunk" section on page 71-1. Click the name of the trunk that you want to update. Continue with Step 6.

Step 3 From the Trunk Type drop-down list, choose the type of trunk.

Step 4 If applicable, from the Device Protocol drop-down list, choose the device protocol.

Step 5 Click **Next**.

Step 6 On the Trunk Configuration window that displays, enter the appropriate settings for gatekeeper-controlled H.225 trunks, gatekeeper-controlled intercluster trunks, and non-gatekeeper-controlled intercluster trunks as described in Table 71-1. For SIP trunks, enter the appropriate settings as described in Table 71-2.

Step 7 To add the new trunk, click **Save**.

The trunk gets added to the database.

If you are updating an existing trunk, click **Reset Trunk** to reset or restart the trunk and apply the new settings.

Note Resetting a trunk **drops** any calls in progress that are using that trunk. Restarting a gateway tries to preserve the calls in progress that are using that gateway, if possible. Other devices wait until calls complete before restarting or resetting. Resetting/restarting an H.323 or SIP device does not physically reset/restart the hardware; it only reinitializes the configuration that is loaded by Cisco Unified CallManager

Additional Topics

See the "Related Topics" section on page 71-28.

Trunk Configuration Settings

Table 71-1 describes the trunk configuration settings for gatekeeper-controlled H.225 trunks, gatekeeper-controlled intercluster trunks, and non-gatekeeper-controlled intercluster trunks.

Table 71-2 describes the trunk configuration settings for SIP trunks.

For more information about related procedures, see the "Related Topics" section on page 71-28.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks

Field	Description
Device Information	
Device Name	Enter a unique identifier for the trunk.
Description	Enter a descriptive name for the trunk.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Device Pool	Choose the appropriate device pool for the trunk. For trunks, device pools specify a list of Cisco Unified CallManagers that the trunk uses to distribute the call load dynamically. **Note** Calls that are initiated from a phone that is registered to a Cisco Unified CallManager that does not belong to the trunk's device pool use different Cisco Unified CallManagers of this device pool for different outgoing calls. Selection of nodes occurs in a random order. A call that is initiated from a phone that is registered to a Cisco Unified CallManager that does belong to the trunk's device pool uses the same Cisco Unified CallManager node for outgoing calls if the Cisco Unified CallManager is up and running.
Call Classification	This parameter determines whether an incoming call through this trunk is considered off the network (OffNet) or on the network (OnNet). When the Call Classification field is configured as Use System Default, the setting of the Cisco Unified CallManager clusterwide service parameter, Call Classification, determines whether the trunk is OnNet or OffNet. This field provides an OnNet or OffNet alerting tone when the call is OnNet or OffNet, respectively. The alerting tones are provided by Cisco Unified CallManager Annunciators. Use this parameter in conjunction with the settings on the Route Pattern Configuration window to classify an outgoing call as OnNet or OffNet.
Media Resource Group List	This list provides a prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music On Hold server, from among the available media resources according to the priority order that a Media Resource Group List defines.
Location	Choose the appropriate location for the trunk. The location specifies the total bandwidth that is available for calls between this location and the central location, or hub. A location setting of Hub_None specifies unlimited available bandwidth.
AAR Group	Choose the automated alternate routing (AAR) group for this device. The AAR group provides the prefix digits that are used to route calls that are otherwise blocked due to insufficient bandwidth. An AAR group setting of None specifies that no rerouting of blocked calls will be attempted.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Tunneled Protocol	This drop-down list box displays for H.225 trunks, gatekeeper-controlled trunks, and non-gatekeeper-controlled trunks. Choose the **QSIG** option if you want to use trunks to transport (tunnel) non-H.323 protocol information in H.323 signaling messages from Cisco Unified CallManager to other Annex M.1-compliant H.323 PINXs. QSIG tunneling supports the following features: Call Completion, Call Diversion, Call Transfer, Identification Services, and Message Waiting Indication.
Packet Capture Mode	This setting exists for troubleshooting encryption only; packet capturing may cause high CPU usage or call-processing interruptions. Choose one of the following options from the drop-down list box: • **None**—This option, which serves as the default setting, indicates that no packet capturing is occurring. After you complete packet capturing, configure this setting. • **Batch Processing Mode**—Cisco Unified CallManager writes the decrypted or nonencrypted messages to a file, and the system encrypts each file. On a daily basis, the system creates a new file with a new encryption key. Cisco Unified CallManager, which stores the file for seven days, also stores the keys that encrypt the file in a secure location. Cisco Unified CallManager stores the file in the PktCap virtual directory. A single file contains the time stamp, source IP address, source IP port, destination IP address, packet protocol, message length, and the message. The IREC tool uses HTTPS, administrator username and password, and the specified day to request a single encrypted file that contains the captured packets. Likewise, the tool requests the key information to decrypt the encrypted file. **Tip** You do not have to reset the trunk after enabling/disabling Packet Capturing. For more information on capturing packets, refer to the *Troubleshooting Guide for Cisco Unified CallManager*.
Packet Capture Duration	This setting exists for troubleshooting encryption only; packet capturing may cause high CPU usage or call-processing interruptions. This field specifies the maximum number of minutes that is allotted for one session of packet capturing. The default setting equals 0, although the range exists from 0 to 300 minutes. To initiate packet capturing, enter a value other than 0 in the field. After packet capturing completes, the value, 0, displays. For more information on capturing packets, refer to the *Cisco Unified CallManager Troubleshooting Guide*.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Media Termination Point Required	This check box is used to indicate whether a media termination point (MTP) is used to implement features that H.323 does not support (such as hold and transfer). Check the Media Termination Point Required check box if you want to use a media termination point to implement features. Uncheck the Media Termination Point Required check box if you do not want to use a media termination point to implement features. Use this check box only for H.323 clients and those H.323 devices that do not support the H.245 Empty Capabilities Set or if you want media streaming to terminate through a single source. If you check this check box to require an MTP and one or both parties are a video endpoint, the call operates as audio only.
Retry Video Call as Audio	This check box applies only to video endpoints that receive a call. For trunks, this check box pertains to calls that are received from Cisco Unified CallManager but not to calls that are received from the wide-area network (WAN). By default, the system checks this check box to specify that this device should immediately retry a video call as an audio call (if it cannot connect as a video call) prior to sending the call to call control for rerouting. If you uncheck this check box, a video call that fails to connect as video does not try to establish as an audio call. The call then fails to call control, and call control routes the call via Automatic Alternate Routing (AAR) and/or route/hunt list.
Wait for Far-End H.245 Terminal Capability Set (H.225 trunks only)	This field applies only to H.323 devices. This check box specifies that Cisco Unified CallManager waits to receive the far-end H.245 Terminal Capability Set before it sends its H.245 Terminal Capability Set. By default, the system checks this check box. To specify that Cisco Unified CallManager should initiate capabilities exchange, uncheck this check box.
Path Replacement Support	If you choose the QSIG option from the Tunneled Protocol drop-down list box, this check box displays for H.225 trunks, gatekeeper-controlled trunks, and non-gatekeeper-controlled trunks. This setting works with QSIG tunneling (Annex M.1) to ensure that non-H.323 information gets sent on the leg of the call that uses path replacement. **Note** The default setting leaves the check box unchecked. When you choose the QSIG Tunneled Protocol option, the system automatically checks the check box.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Transmit UTF-8 for Calling Party Name	This device uses the user locale setting of the SIP trunks to determine whether to send Unicode and whether to translate received Unicode information. For the sending device, if you check this check box and the user locale setting in the device's device pool matches the terminating phone's user locale, the device sends Unicode. If the user locale settings do not match, the device sends ASCII. The receiving device translates incoming Unicode characters based on the user locale setting of the sending device's device pool. If the user locale setting matches the terminating phone's user locale, the phone displays the characters. **Note** The phone may display garbled characters if the two ends of the trunk configure user locales that do not belong to the same language group.
Unattended Port	Check this check box if calls can be redirected, transferred and forwarded to an unattended port, such as a voice mail port. The default value for this check box leaves it unchecked.
SRTP Allowed	Check the SRTP Allowed check box if you want Cisco Unified CallManager to allow secure and nonsecure calls over the trunk. If you do not check this check box, Cisco Unified CallManager prevents SRTP negotiation with the trunk and uses RTP. **Caution** If you check this check box, Cisco strongly recommends that you configure IPSec, so you do not expose keys and other security-related information during call negotiations. If you do not configure IPSec correctly, consider signaling between Cisco Unified CallManager and the gateway as nonsecure. For more information on encryption for trunks, refer to the *Cisco Unified CallManager Security Guide*.
Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain	From the drop-down list box, choose an MLPP domain to associate with this device. If you leave this field blank, this device inherits its MLPP domain from the value that was set for the device pool. If the device pool does not have an MLPP Domain setting, this device inherits its MLPP Domain from the value that was set for the MLPP Domain Identifier enterprise parameter.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
MLPP Indication	If available, this setting specifies whether a device that is capable of playing precedence tones will use the capability when it places an MLPP precedence call. From the drop-down list box, choose a setting to assign to this device from the following options: • Default-This device inherits its MLPP indication setting from its device pool. • Off-This device does not handle nor process indication of an MLPP precedence call. • On-This device does handle and process indication of an MLPP precedence call. **Note** Do not configure a device with the following combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.
Call Routing Information	
Inbound Calls	
Significant Digits	Significant digits represent the number of final digits that are retained on inbound calls. Use for the processing of incoming calls and to indicate the number of digits that are used to route calls that are coming in to the H.323 device. Choose the number of significant digits to collect, from 0 to 32. Cisco Unified CallManager counts significant digits from the right (last digit) of the number that is called.
Calling Search Space	From the drop-down list box, choose the appropriate calling search space for the trunk. The calling search space specifies the collection of route partitions that are searched to determine how to route a collected (originating) number. You can configure the number of calling search spaces that display in this drop-down list box by using the Max List Box Items enterprise parameter. If more calling search spaces exist than the number that are configured in the Max List Box Items enterprise parameter, the ellipsis button (**...**) displays next to the drop-down list box. Click the "**...**" button to display the Select Calling Search Space window. Enter a partial calling search space name in the **List items where Name contains** field. Click the desired calling search space name in the list of calling search spaces that displays in the **Select item to use** box and click **OK**. **Note** To set the maximum list box items, choose **System > Enterprise Parameters** and choose **CCMAdmin Parameters**.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
AAR Calling Search Space	Choose the appropriate calling search space for the device to use when performing automated alternate routing (AAR). The AAR calling search space specifies the collection of route partitions that are searched to determine how to route a collected (originating) number that is otherwise blocked due to insufficient bandwidth.
Prefix DN	Enter the prefix digits that are appended to the called party number on incoming calls. Cisco Unified CallManager adds prefix digits after first truncating the number in accordance with the Significant Digits setting.
Redirecting Number IE Delivery - Inbound	Check this check box to accept the Redirecting Number IE. (The Redirecting Number IE gets sent in the UUIE.) Uncheck the check box to exclude the Redirecting Number IE. You use Redirecting Number IE for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number IE, you should check the check box. **Note** Default leaves the check box checked. You cannot check this check box if you choose the QSIG option from the Tunneled Protocol drop-down list box.
Enable Inbound FastStart	Check this check box to enable the H.323 FastStart call connections on incoming calls. By default, the check box remains unchecked for the H.323 gateway. For intercluster calls, you must check the Enable Inbound FastStart check box on Cisco Unified CallManager servers in other clusters for the outbound FastStart feature to work. If you updated Cisco CallManager 3.3(2) servers in other clusters with support patch B, do not enable inbound FastStart because 3.3(2)spB does not support the inbound FastStart feature over intercluster trunks
Outbound Calls	
Calling Party Selection	Choose the directory number that is sent on an outbound call on a gateway. The following options specify which directory number is sent: • Originator—Send the directory number of the calling device. • First Redirect Number—Send the directory number of the redirecting device. • Last Redirect Number—Send the directory number of the last device to redirect the call. • First Redirect Number (External)—Send the external directory number of the redirecting device. • Last Redirect Number (External)—Send the external directory number of the last device to redirect the call.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Calling Line ID Presentation	Cisco Unified CallManager uses calling line ID presentation (CLIP) as a supplementary service to control the display of the calling party's number on the called party's phone display screen. Choose Default if you do not want to change the presentation setting. Choose Allowed if you want calling number information to display. Choose Restricted if you do not want the calling number information to display.
Called Party IE Number Type Unknown	Choose the format for the type of number in called party directory numbers. Cisco Unified CallManager sets the called directory number (DN) type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the called directory number to be encoded to a non-national numbering plan type. Choose one of the following options: • Cisco Unified CallManager—Cisco Unified CallManager sets the directory number type. • Unknown—This option indicates that the dialing plan is unknown. • National—Use when you are dialing within the dialing plan for your country. • International—Use when you are dialing outside the dialing plan for your country. • Subscriber—Use when you are dialing a subscriber by using a shortened subscriber number.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Calling Party IE Number Type Unknown	Choose the format for the type of number in calling party directory numbers. Cisco Unified CallManager sets the calling directory number (DN) type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the calling directory number to be encoded to a non-national numbering plan type. Choose one of the following options: • Cisco Unified CallManager—Cisco Unified CallManager sets the directory number type. • Unknown—This option indicates that the dialing plan is unknown. • National—Use when you are dialing within the dialing plan for your country. • International—Use when you are dialing outside the dialing plan for your country. • Subscriber—Use when you are dialing a subscriber by using a shortened subscriber number.
Called Numbering Plan	Choose the format for the numbering plan in called party directory numbers. Cisco Unified CallManager sets the called DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the called numbering plan to be encoded to a non-national numbering plan. Choose one of the following options: • Cisco Unified CallManager—Cisco Unified CallManager sets the Numbering Plan in the directory number. • ISDN—Use when you are dialing outside the dialing plan for your country. • National Standard—Use when you are dialing within the dialing plan for your country. • Private—Use when you are dialing within a private network. • Unknown—This option indicates that the dialing plan is unknown.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Calling Numbering Plan	Choose the format for the numbering plan in calling party directory numbers. Cisco Unified CallManager sets the calling DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the calling numbering plan to be encoded to a non-national numbering plan. Choose one of the following options: • Cisco Unified CallManager—Cisco Unified CallManager sets the Numbering Plan in the directory number. • ISDN—Use when you are dialing outside the dialing plan for your country. • National Standard—Use when you are dialing within the dialing plan for your country. • Private—Use when you are dialing within a private network. • Unknown—This option indicates that the dialing plan is unknown.
Caller ID DN	Enter the pattern, from 0 to 24 digits, that you want to use to format the caller ID on outbound calls from the trunk. For example, in North America • 555XXXX = Variable Caller ID, where X represents an extension number. The Central Office (CO) appends the number with the area code if you do not specify it. • 5555000 = Fixed Caller ID. Use this form when you want the Corporate number to be sent instead of the exact extension from which the call is placed. The CO appends the number with the area code if you do not specify it.
Display IE Delivery	Check this check box to enable delivery of the display information element (IE) in SETUP and CONNECT messages for the calling and called party name delivery service. **Note** The default setting leaves this check box checked. You cannot check this check box if you choose the QSIG option from the Tunneled Protocol drop-down list box.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Redirecting Number IE Delivery - Outbound	Check this check box to transmit the first Redirecting Number and the redirecting reason of the call when the call is forwarded. (The Redirecting Number IE gets sent in the UUIE.) Uncheck the check box to exclude the first Redirecting Number and the redirecting reason. You use Redirecting Number IE for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number IE, you should check the check box. **Note** The default setting leaves this check box checked. You cannot check this check box if you choose the QSIG option from the Tunneled Protocol drop-down list box.
Enable Outbound FastStart	Check this check box to enable the H.323 FastStart feature on outgoing calls. By default, the check box remains unchecked for the H.323 gateway or trunk. When you check the Enable Outbound FastStart check box, you must set the Media Termination Point Required, Media Resource Group Lists, and Codec for Outbound FastStart.
Codec For Outbound FastStart	Choose the codec for use with the H.323 device for an outbound FastStart call: • G711 mu-law 64K (default) • G711 a-law 64K • G723 • G729 • G729AnnexA • G729AnnexB • G729AnnexA-AnnexB When you check the Enable Outbound FastStart check box, you must choose the codec for supporting outbound FastStart calls.
Gatekeeper Information (for gatekeeper-controlled H.225 trunks and intercluster trunks)	
Gatekeeper Name	Choose the gatekeeper that controls this trunk. **Note** For a gatekeeper-controlled trunk to register correctly with a gatekeeper through use of H.323 dynamic addressing, you must set the Send Product ID and Version ID service parameter to *True*. (The default value specifies *False*.) To do so, choose **System > Service Parameters** and find the Send Product ID and Version ID service parameter for the Cisco CallManager service in the Clusterwide Parameters (Device - H323) portion of the Service Parameter Configuration window.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Terminal Type	Use the Terminal Type field to designate the type for all devices that this trunk controls. Always set this field to Gateway for normal trunk call admission control.
Technology Prefix	Use this optional field to eliminate the need for entering the IP address of every Cisco Unified CallManager when configuring the **gw-type-prefix** on the gatekeeper: • If you leave this field blank (the default setting), you must specify the IP address of each Cisco Unified CallManager that can register with the gatekeeper when you enter the **gw-type-prefix** command on the gatekeeper. • When you use this field, make sure that the value that you enter exactly matches the *type-prefix* value that is specified with the **gw-type-prefix** command on the gatekeeper. For example, if you leave this field blank and you have two Cisco Unified CallManagers with IP addresses of 10.1.1.2 and 11.1.1.3, enter the following **gw-type-prefix** command on the gatekeeper: **gw-type-prefix 1#* default-technology gw ip 10.1.1.2 gw ip 11.1.1.3** If you enter **1#*** in this field, enter the following **gw-type-prefix** command on the gatekeeper: **gw-type-prefix 1#* default-technology**
Zone	Use this optional field to request a specific zone on the gatekeeper with which Cisco Unified CallManager will register. The zone specifies the total bandwidth that is available for calls between this zone and another zone: • If you do not enter a value in this field, the **zone subnet** command on the gatekeeper determines the zone with which Cisco Unified CallManager registers. Cisco recommends the default setting for most configurations. • If you want Cisco Unified CallManager to register with a specific zone on the gatekeeper, enter the value in this field that exactly matches the zone name that is configured on the gatekeeper with the **zone** command. Specifying a zone name in this field eliminates the need for a **zone subnet** command for each Cisco Unified CallManager that is registered with the gatekeeper. Refer to the command reference documentation for your gatekeeper for more information.
Remote Cisco Unified CallManager Information (for non-gatekeeper-controlled intercluster trunks)	
Server 1 IP Address/Host Name	Enter the IP address or host name of the first remote Cisco Unified CallManager that this trunk accesses.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
Server 2 IP Address/Host Name	Enter the IP address or host name of the second remote Cisco Unified CallManager that this trunk accesses. **Note** If this non-gatekeeper-controlled intercluster trunk accesses the device pool of a remote non-gatekeeper-controlled intercluster trunk and that device pool has a second Cisco Unified CallManager node, you must enter the second remote Cisco Unified CallManager IP address/host name in this field.
Server 3 IP Address/Host Name	Enter the IP address or host name of the third remote Cisco Unified CallManager that this trunk accesses. **Note** If this non-gatekeeper-controlled intercluster trunk accesses the device pool of a remote non-gatekeeper-controlled intercluster trunk and that device pool has a third Cisco Unified CallManager node, you must enter the third remote Cisco Unified CallManager IP address/host name in this field.
MLPP Indication	If available, this setting specifies whether a device that is capable of playing precedence tones will use the capability when it places an MLPP precedence call. From the drop-down list box, choose a setting to assign to this device from the following options: • **Default**—This device inherits its MLPP indication setting from its device pool. • **Off**—This device does not handle nor process indication of an MLPP precedence call. • **On**—This device does handle and process indication of an MLPP precedence call. **Note** Do not configure a device with the following combination of settings: MLPP Indication is set to *Off or Default* (when default is *Off*) while MLPP Preemption is set to *Forceful*.

Table 71-1 Trunk Configuration Settings for H.225 and Intercluster Trunks (continued)

Field	Description
MLPP Preemption (not available for H.323 ICT)	If available, this setting specifies whether a device that is capable of preempting calls in progress will use the capability when it places an MLPP precedence call. From the drop-down list box, choose a setting to assign to this device from the following options: • **Default**—This device inherits its MLPP preemption setting from its device pool. • **Disabled**—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls. • **Forceful**—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls. **Note** Do not configure a device with the following combination of settings: MLPP Indication is set to *Off or Default* (when default is *Off*) while MLPP Preemption is set to *Forceful*. **Note** The Trunk page currently does not make the MLPP Preemption flag available. The location-based MLPP preemption flag controls the preemption logic.

Table 71-2 describes the trunk configuration settings for SIP trunks.

Table 71-2 Trunk Configuration Settings for SIP Trunks

Field	Description
Device Information	
Device Name	Enter a unique identifier for the trunk.
Description	Enter a descriptive name for the trunk.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Device Pool	Choose the appropriate device pool for the trunk. For trunks, device pools specify a list of Cisco Unified CallManagers that the trunk uses to distribute the call load dynamically. **Note** Calls that are initiated from a phone that is registered to a Cisco Unified CallManager that does not belong to the trunk's device pool use different Cisco Unified CallManagers of this device pool for different outgoing calls. Selection of Cisco Unified CallManager nodes occurs in a random order. A call that is initiated from a phone that is registered to a Cisco Unified CallManager that does belong to the trunk's device pool uses the same Cisco Unified CallManager node for outgoing calls if the Cisco Unified CallManager is up and running. The default value for Device Pool specifies Not Selected.
Call Classification	This parameter determines whether an incoming call through this trunk is considered off the network (OffNet) or on the network (OnNet). The default value for Call Classification is Use System Default. When the Call Classification field is configured as Use System Default, the setting of the Cisco Unified CallManager clusterwide service parameter, Call Classification, determines whether the trunk is OnNet or OffNet. This field provides an OnNet or OffNet alerting tone when the call is OnNet or OffNet, respectively. Use this parameter in conjunction with the settings on the Route Pattern Configuration window to classify an outgoing call as OnNet or OffNet.
Media Resource Group List	This list provides a prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music On Hold server, from among the available media resources according to the priority order that a Media Resource Group List defines. The default value for Media Resource Group List specifies None.
Location	Choose the appropriate location for the trunk. The location specifies the total bandwidth that is available for calls between this location and the central location, or hub. A location setting of Hub_None specifies unlimited available bandwidth. The location also associates with the RSVP policy with regard to other locations. The configuration allows RSVP to be enabled and disabled based upon location pairs. The default value for Location specifies Hub_None.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
AAR Group	Choose the automated alternate routing (AAR) group for this device. The AAR group provides the prefix digits that are used to route calls that are otherwise blocked due to insufficient bandwidth. An AAR group setting of None specifies that no rerouting of blocked calls will be attempted. The default value for AAR Group specifies None.
Packet Capture Mode	This setting exists for troubleshooting encryption only; packet capturing may cause high CPU usage or call-processing interruptions. Choose one of the following options from the drop-down list box: • **None**—This option, which serves as the default setting, indicates that no packet capturing is occurring. After you complete packet capturing, configure this setting. • **Batch Processing Mode**—Cisco Unified CallManager writes the decrypted or nonencrypted messages to a file, and the system encrypts each file. On a daily basis, the system creates a new file with a new encryption key. Cisco Unified CallManager, which stores the file for seven days, also stores the keys that encrypt the file in a secure location. Cisco Unified CallManager stores the file in the PktCap virtual directory. A single file contains the time stamp, source IP address, source IP port, destination IP address, packet protocol, message length, and the message. The TAC debugging tool uses HTTPS, administrator username and password, and the specified day to request a single encrypted file that contains the captured packets. Likewise, the tool requests the key information to decrypt the encrypted file. Before you contact TAC, you must capture the SRTP packets by using a sniffer trace between the affected devices. For more information on capturing packets, refer to the *Troubleshooting Guide for Cisco Unified CallManager*.
Packet Capture Duration	This setting exists for troubleshooting encryption only; packet capturing may cause high CPU usage or call-processing interruptions. This field specifies the maximum number of minutes that is allotted for one session of packet capturing. The default setting equals 0, although the range exists from 0 to 300 minutes. To initiate packet capturing, enter a value other than 0 in the field. After packet capturing completes, the value, 0, displays. For more information on capturing packets, refer to the *Cisco Unified CallManager Troubleshooting Guide*.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Media Termination Point Required	You can configure Cisco Unified CallManager SIP trunks to always use an MTP. Check this check box to provide media channel information in the outgoing INVITE request. When this check box is checked, all media channels must terminate and reoriginate on the MTP device. If you uncheck the check box, the Cisco Unified CallManager can decide whether calls are to go through the MTP device or be connected directly between the endpoints. **Note** If check box remains unchecked (default case), Cisco Unified CallManager will attempt to dynamically allocate an MTP if the DTMF methods for the call legs are not compatible. For example, existing SCCP phones support only out-of-band DTMF, and existing SIP phones support RFC2833. Because the DTMF methods are not identical, the Cisco Unified CallManager will dynamically allocate an MTP. If, however, a new SCCP phone, which supports RFC2833 and out-of-band, calls an existing SIP phone, Cisco Unified CallManager will not allocate an MTP because both phones support RFC2833. So, by having the same type of DTMF method supported on each phone, no need exists for MTP.
Retry Video Call as Audio	This check box pertains to outgoing SIP trunk calls and does not impact incoming calls. By default, the system checks this check box to specify that this device should immediately retry a video call as an audio call (if it cannot connect as a video call) prior to sending the call to call control for rerouting. If you uncheck this check box, a video call that fails to connect as video does not try to establish as an audio call. The call then fails to call control, and call control routes the call via Automatic Alternate Routing (AAR) and/or route/hunt list.
Transmit UTF-8 for Calling Party Name	This device uses the user locale setting of the device pool to determine whether to send Unicode and whether to translate received Unicode information. For the sending device, if you check this check box and the user locale setting in the device pool matches the terminating phone user locale, the device sends Unicode. If the user locale settings do not match, the device sends ASCII. The receiving device translates incoming Unicode characters based on the user locale setting of the sending device pool. If the user locale setting matches the terminating phone's user locale, the phone displays the characters. **Note** The phone may display malformed characters if the two ends of the trunk configure user locales that do not belong to the same language group. The default value for Transmit UTF-8 for Calling Party Name leaves the check box unchecked.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Unattended Port	Check this check box if calls can be redirected and transferred to an unattended port, such as a voice mail port. The default value for this check box leaves it unchecked.
Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain	From the drop-down list, choose an MLPP domain to associate with this device. If you leave this field blank, this device inherits its MLPP domain from the value that is set for the device pool. If the device pool does not have an MLPP Domain setting, this device inherits its MLPP Domain from the value that is set for the MLPP Domain Identifier enterprise parameter. The default value for MLPP Domain specifies None.
Call Routing Information	
Inbound Calls	
Significant Digits	Significant digits represent the number of final digits that are retained on inbound calls. Use for the processing of incoming calls and to indicate the number of digits that are used to route calls that are coming in to the SIP device. Choose the number of significant digits to collect, from 0 to 32, or choose All. **Note** Cisco Unified CallManager counts significant digits from the right (last digit) of the number that is called. The default value for Significant Digits specifies All.
Connected Line ID Presentation	Cisco Unified CallManager uses connected line ID presentation (COLP) as a supplementary service to provide the calling party with the connected party number. The SIP trunk level configuration takes precedence over the call-by-call configuration. The default value for Connected Line ID Presentation specifies Default, which translates to Allowed.Choose Default if you want Cisco Unified CallManager to send connected line information. Choose Restricted if you do not want Cisco Unified CallManager to send connected line information.
Connected Name Presentation	Cisco Unified CallManager uses connected name ID presentation (CONP) as a supplementary service to provide the calling party with the connected party's name. The SIP trunk level configuration takes precedence over the call-by-call configuration. The default value for Connected Name Presentation specifies Default, which translates to Allowed. Choose Default if you want Cisco Unified CallManager to send connected name information. Choose Restricted if you do not want Cisco Unified CallManager to send connected name information.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Calling Search Space	From the drop-down list box, choose the appropriate calling search space for the trunk. The calling search space specifies the collection of route partitions that are searched to determine how to route a collected (originating) number. You can configure the number of calling search spaces that display in this drop-down list box by using the Max List Box Items enterprise parameter. If more calling search spaces exist than the number that is configured in the Max List Box Items enterprise parameter, the ellipsis button (**...**) displays next to the drop-down list box. Click the "**...**" button to display the Select Calling Search Space window. Enter a partial calling search space name in the **List items where Name contains** field. Click the desired calling search space name in the list of calling search spaces that displays in the **Select item to use** box and click **OK**. **Note** To set the maximum list box items, choose **System > Enterprise Parameters** and choose **CCMAdmin Parameters**. The default value for Calling Search Space specifies None.
AAR Calling Search Space	Choose the appropriate calling search space for the device to use when performing automated alternate routing (AAR). The AAR calling search space specifies the collection of route partitions that are searched to determine how to route a collected (originating) number that is otherwise blocked due to insufficient bandwidth. The default value for AAR Calling Search Space specifies None.
Prefix DN	Enter the prefix digits that are appended to the called party number on incoming calls. Cisco Unified CallManager adds prefix digits after first truncating the number in accordance with the Significant Digits setting.
Redirecting Diversion Header Delivery - Inbound	Check this check box to accept the Redirecting Number in the incoming INVITE message to the Cisco Unified CallManager. Uncheck the check box to exclude the Redirecting Number in the incoming INVITE message to the Cisco Unified CallManager. You use Redirecting Number for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number, you should check the check box. The default value for Redirecting Number IE Deliver - Inbound specifies not checked.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Outbound Calls	
Calling Party Selection	Choose the directory number that is sent on an outbound call. The following options specify which directory number is sent: • Originator—Send the directory number of the calling device. • First Redirect Number—Send the directory number of the redirecting device. • Last Redirect Number—Send the directory number of the last device to redirect the call. • First Redirect Number (External)—Send the external directory number of the redirecting device. • Last Redirect Number (External)—Send the external directory number of the last device to redirect the call. The default value for Calling Party Selection specifies Originator.
Calling Line ID Presentation	Cisco Unified CallManager uses calling line ID presentation (CLIP) as a supplementary service to provide the calling party number. The SIP trunk level configuration takes precedence over the call-by-call configuration. The default value for Calling Line ID Presentation specifies Default, which translates to Allowed. Choose Default if you want Cisco Unified CallManager to send calling number information. Choose Restricted if you do not want Cisco Unified CallManager to send the calling number information.
Calling Name Presentation	Cisco Unified CallManager uses calling name ID presentation (CNIP) as a supplementary service to provide the calling party name. The SIP trunk level configuration takes precedence over the call-by-call configuration. Choose *Allowed*, which is the default, if you want Cisco Unified CallManager to send calling name information. Choose *Restricted* if you do not want Cisco Unified CallManager to send the calling name information. The default value for Calling Name Presentation specifies Default.
Caller ID DN	Enter the pattern, from 0 to 24 digits, that you want to use to format the caller ID on outbound calls from the trunk. For example, in North America • 555XXXX = Variable Caller ID, where X represents an extension number. The Central Office (CO) appends the number with the area code if you do not specify it. • 5555000 = Fixed Caller ID. Use this form when you want the Corporate number to be sent instead of the exact extension from which the call is placed. The CO appends the number with the area code if you do not specify it.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Caller Name	Check this check box to override the caller name that is received from the originating SIP device.
Redirecting Diversion Header Delivery - Outbound	Check this check box to include the Redirecting Number in the outgoing INVITE message from the Cisco Unified CallManager to indicate the original called party number and the redirecting reason of the call when the call is forwarded. Uncheck the check box to exclude the first Redirecting Number and the redirecting reason from the outgoing INVITE message. You use Redirecting Number for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number, you should check the check box. The default value for Redirecting Number IE Delivery - Outbound specifies check box does not get checked.
SIP Information	
Destination Address	The Destination Address represents the remote SIP peer with which this trunk will communicate. The allowed values for this field specify a valid V4 dotted IP address, fully qualified domain name (FQDN), or DNS SRV record only if the *Destination Address is an SRV* field is checked. **Note** SIP trunks only accept incoming requests from the configured Destination Address and the specified incoming port that is specified in the SIP Trunk Security Profile that is associated with this trunk. If the remote end is a Cisco Unified CallManager cluster, DNS SRV represents the recommended choice for this field. The DNS SRV record should include all Cisco Unified CallManagers within the cluster.
Destination Address is an SRV	This field specifies that the configured Destination Address is an SRV record. The default value specifies unchecked.
Destination Port	Choose the destination port. Ensure that the value that you enter specifies any port from 1024 - 65535. **Note** You can now have the same port number that is specified for multiple trunks. You need not enter a value if the destination address is an DNS SRV port. The default 5060 indicates the SIP port. The default value for Destination Port specifies 5060.
MTP Preferred Originating Codec	Indicate the preferred outgoing codec. This field gets used only when the MTP Termination Point Required check box is checked.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Presence Group	Configure this field with the Presence feature. From the drop-down list box, choose a Presence group for the SIP trunk. The selected group specifies the destinations that the device/application/server that is connected to the SIP trunk can monitor. The default value for Presence Group specifies Standard Presence group, configured with installation. Presence groups that are configured in Cisco Unified CallManager Administration also appear in the drop-down list box. Presence authorization works with presence groups to allow or block presence requests between groups. Refer to the "Presence" chapter in the *Cisco Unified CallManager Features and Services Guide* for information about configuring permissions between groups. **Tip** A presence group can be applied to the SIP trunk or to the application that is connected to the SIP trunk. If a presence group is configured for both a SIP trunk and SIP trunk application, the presence group that is applied to the application overrides the presence group that is applied to the trunk.
SIP Trunk Security Profile	Choose the security profile to apply to the SIP trunk. You must apply a security profile to all SIP trunks that are configured in Cisco Unified CallManager Administration. Installing Cisco Unified CallManager provides a predefined, nonsecure SIP trunk security profile for autoregistration. To enable security features for a SIP trunk, configure a new security profile and apply it to the SIP trunk. If the trunk does not support security, choose a nonsecure profile. To identify the settings that the profile contains, choose **System > Security Profile > SIP Trunk Security Profile.** For information on how to configure security profiles, refer to the *Cisco Unified CallManager Security Guide*. The default value for SIP Trunk Security Profile specifies Not Selected.
Rerouting Calling Search Space	Calling search spaces determine the partitions that calling devices can search when attempting to complete a call. The rerouting calling search space gets used to determine where a SIP user (A) can refer another user (B) to a third party (C). After the refer is completed, B and C connect. In this case, the rerouting calling search space that is used is that of the initial SIP user (A). **Note** Calling Search Space also applies to 3xx redirection and INVITE with Replaces features. The default value for Rerouting Calling Search Space specifies None.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
Out-of-Dialog Refer Calling Search Space	Calling search spaces determine the partitions that calling devices can search when attempting to complete a call. The out-of-dialog calling search space gets used when a Cisco Unified CallManager refers a call (B) coming into SIP user (A) to a third party (C) when there is no involvement of SIP user (A). In this case, the system uses the out-of-dialog calling search space of SIP user (A). The default value for Out-of-Dialog Refer Calling Search Space specifies None.
SUBSCRIBE Calling Search Space	Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified CallManager routes presence requests from the device/server/application that connects to the SIP trunk. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the SIP trunk. From the drop-down list box, choose the SUBSCRIBE calling search space to use for presence requests for the SIP trunk. All calling search spaces that you configure in Cisco Unified CallManager Administration display in the SUBSCRIBE Calling Search Space drop-down list box. If you do not select a different calling search space for the SIP trunk from the drop-down list, the SUBSCRIBE calling search space defaults to None. To configure a SUBSCRIBE calling search space specifically for this purpose, you configure a calling search space as you do all calling search spaces. For information on how to configure a calling search space, see the "Calling Search Space Configuration" section on page 42-1.

Table 71-2 Trunk Configuration Settings for SIP Trunks (continued)

Field	Description
SIP Profile	From the drop-down list box, choose the SIP profile that is to be used for this SIP trunk. The default value for SIP Profile specifies None Selected.
DTMF Signaling Method	Choose from the following options: No Preference (default)—Cisco Unified CallManager will pick the DTMF method to negotiate DTMF, so an MTP is not required for the call. If Cisco Unified CallManager has no choice but to allocate an MTP (if the Media Termination Point Required check box is checked), SIP trunk will negotiate DTMF to RFC2833. RFC 2833—Choose this configuration if the preferred DTMF method to be used across the trunk is RFC2833. Cisco Unified CallManager makes every effort to negotiate RFC2833 regardless of MTP usage. Out of band provides the fallback method if the peer endpoint supports it. OOB and RFC 2833—Choose this configuration if both out of band and RFC2833 should be used for DTMF. **Note** If the peer endpoint supports both out of band and RFC2833, Cisco Unified CallManager will negotiate both out-of-band and RFC2833 DTMF methods. As a result, two DTMF events would get sent for the same DTMF keypress (one out of band and the other, RFC2833).

Additional Topics

See the "Related Topics" section on page 71-28.

Deleting a Trunk

Perform the following steps to delete a trunk.

Before You Begin

You cannot delete a trunk that is assigned to one or more route patterns. To find out which route patterns are using the trunk, in the Trunk Configuration window, choose **Dependency Records** from the Related Links drop-down list box and click **Go**. If dependency records are not enabled for the system, the Dependency Records Summary window displays a message. For more information about dependency records, see the "Accessing Dependency Records" section on page A-2. If you try to delete a trunk that is in use, Cisco Unified CallManager displays a message. Before deleting a trunk that is currently in use, you must perform either or both of the following tasks:

- Assign a different trunk to any route patterns that are using the trunk that you want to delete. See the "Configuring a Route Pattern" section on page 34-3.
- Delete the route patterns that are using the trunk that you want to delete. See the "Deleting a Route Pattern" section on page 34-10.

Procedure

Step 1 Choose **Device** > **Trunk**.

The Find and List Trunks window displays.

Step 2 To locate a specific trunk, enter search criteria and click **Find**.

A list of trunks that match the search criteria displays.

Step 3 Perform one of the following actions:

- Check the check boxes next to the trunks that you want to delete and click **Delete Selected**.
- Delete all trunks in the window by clicking **Select All** and then clicking **Delete Selected**.
- From the list, choose the name of the trunk that you want to delete to display its current settings and click **Delete**.

A confirmation dialog displays.

Step 4 To delete the trunk, click **OK**.

Additional Topics

See the "Related Topics" section on page 71-28.

Resetting a Trunk

Perform the following procedure to reset the trunk.

Caution Resetting devices can cause them to drop calls.

Procedure

Step 1 Choose **Device > Trunk**.

The Find and List Trunks window displays.

Step 2 To locate a specific trunk, enter search criteria and click **Find**.

A list of trunks that match the search criteria displays.

Step 3 From the list, click the name of the trunk that you want to reset.

The Trunk Configuration window displays.

Step 4 After you change any settings for the Trunk Device, click **Reset Trunk.**

The Device Reset dialog displays.

Step 5 Click one of the following choices:

- **Restart**—Restarts the trunk device without shutting it down first.
- **Reset**—Shuts down, then restarts the internal trunk device. The Cisco Unified CallManager cluster unregisters (URQ) and then reregisters (RRQ) with the trunk if the trunk is gatekeeper controlled.
- **Close**—Closes the Reset Device dialog without performing any action.

Note For SIP trunks, Restart and Reset behave the same way, so all active calls will disconnect when either choice is pressed. Trunks do not have to undergo a Restart or Reset when Packet Capture is enabled or disabled.

Additional Topics

See the "Related Topics" section on page 71-28.

Related Topics

- Finding a Trunk, page 71-1
- Configuring a Trunk, page 71-2
- Trunk Configuration Settings, page 71-3
- Deleting a Trunk, page 71-26
- Resetting a Trunk, page 71-27
- Configuring SIP Profiles, page 79-2
- SIP Trunk Security Profile Configuration
- Call Admission Control, *Cisco Unified CallManager System Guide*
- Gatekeepers and Trunks, *Cisco Unified CallManager System Guide*
- Gatekeeper and Trunk Configuration in Cisco Unified CallManager, *Cisco Unified CallManager System Guide*
- *Cisco Unified CallManager Security Guide*
- *Cisco Unified Communications Solution Reference Network Design (SRND)*