System Configuration Guide for Cisco Unified Communications Manager, Release 11.5(1)SU1

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System Configuration Overview

- About System Configuration, on page 1
- Configuration Tools Overview, on page 1
- System Configuration High-Level Flow, on page 3

About System Configuration

This document provides information about the tasks that you need to perform in order to configure the call control system. It contains information such as task flows, procedures, and prerequisites.


Configuration Tools Overview

The procedures in this guide require you to use the following two configuration tools:

- Cisco Unified Communications Manager Administration
- Cisco Unified Serviceability

This chapter provides a brief description of the tools and how to access them.

Cisco Unified Communications Manager Administration

Cisco Unified Communications Manager Administration Administration is a web-based application that allows you to make individual, manual configuration changes to the Unified Communications Manager nodes. The procedures in this guide describe how to configure features using this application.

If you need to perform bulk configuration tasks and want to automate the configuration process, you can use the Unified Communications Manager Bulk Administration Tool (BAT) to make a large number of configuration changes at the same time. For more information, see Bulk Administration Guide for Cisco Unified Communications Manager.
Log In to Cisco Unified CM Administration

Use the following procedure to log in to Cisco Unified Communications Manager Administration. After you log in to Cisco Unified Communications Manager Administration, messages may display that indicate the current state of licenses for Unified Communications Manager in the main window. For example, Unified Communications Manager may identify the following situations:

- Unified Communications Manager currently operates with starter (demo) licenses, so upload the appropriate license files.
- Unified Communications Manager currently operates with an insufficient number of licenses, so upload additional license files.
- Unified Communications Manager does not currently use the correct software feature license. In this case, the Cisco CallManager service stops and does not start until you upload the appropriate software version license and restart the Cisco CallManager service.

Log In to Cisco Unified CM Administration

Use the following procedure to browse into the server and log in to Cisco Unified CM Administration.

Procedure

Step 1
Start your preferred operating system browser.

Step 2
In the address bar of the web browser, enter the following case-sensitive URL:
https://<Unified CM-server-name>:{8443}/ccmadmin/showHome.do

where: <Unified CM-server-name> equals the name or IP address of the server

Note  You can optionally specify a port number.

Step 3
A Security Alert dialog box displays. Click the appropriate button.

Step 4
At the main Cisco Unified CM Administration window, enter the username and password that you specified during Unified Communications Manager installation and click Login. (If you want to clear the content of both fields, click Reset.)

Note  For security purposes, Cisco Unified Communications Manager Administration logs you out after 30 minutes of inactivity, and you must log back in.

Cisco Unified Communications Manager Serviceability

Some procedures in this guide require you to use the Cisco Unified Serviceability application to start or restart services on the Unified Communications Manager nodes.

Cisco Unified Serviceability is a web-based troubleshooting tool that provides the following functionality:

- Saves alarms and events for troubleshooting and provides alarm message definitions.
- Saves trace information to log files for troubleshooting.
- Monitors real-time behavior of components through the Cisco Unified Real-Time Monitoring Tool (Unified RTMT).
• Provides audit capability by logging configuration changes to the system by a user or due to result of the user action. This functionality supports the Information Assurance feature of Unified Communications Manager and Cisco Unity Connection.

• Provides feature services that you can activate, deactivate, and view through the Service Activation window.

• Generates and archives daily reports; for example, alert summary or server statistic reports.

• Allows Unified Communications Manager, Instant Messaging and Presence and Cisco Unity Connection to work as a managed device for Simple Network Management Protocol (SNMP) remote management and troubleshooting.

• Monitors the disk usage of the log partition on a node (or all nodes in the cluster).

• Monitors the number of threads and processes in the system; uses cache to enhance the performance.

• Unified Communications Manager only: Generates Unified Communications Manager reports for Quality of Service, traffic, and billing information through Cisco Unified Communications Manager CDR Analysis and Reporting.

Log into Cisco Unified Communications Manager Serviceability

Use the following procedure to log in to Cisco Unified Serviceability.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Start your preferred operating system browser.</td>
</tr>
</tbody>
</table>
| Step 2 | In the address bar of the web browser, enter the following case-sensitive URL:  
https://<Unified CM-server-name>:{8443}/ccmadmin/showHome.do  
where: <Unified CM-server-name> equals the name or IP address of the server |
| Step 3 | A Security Alert dialog box displays. Click the appropriate button. |
| Step 4 | From Cisco Unified CM Administration, choose Cisco Unified Serviceability from the Navigation menu drop-down list and click Go. |
| Step 5 | Enter the username and password that you specified during Unified Communications Manager installation and click Login. |

**Note**  
For security purposes, the system logs you out after 30 minutes of inactivity, and you must log back in.

System Configuration High-Level Flow

Complete your system configuration in the following order. The steps link to sections with configurations for a group of common system components.
## System Configuration Guide for Cisco Unified Communications Manager, Release 11.5(1)SU1

### System Configuration Overview

#### System Configuration High-Level Flow

## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Initial Configuration Task Flow, on page 9</td>
<td>Complete the configurations in this part of the guide to configure the initial parameters for the system, such as licensing, server information, device pools, and port settings.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Inbound and Outbound Calling Configuration, on page 61</td>
<td>Complete the configurations in this part of the guide to configure gateways and survivable remote telephony systems.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Dial Plan Configuration, on page 111</td>
<td>Complete the configurations in this part of the guide to configure the elements of the dial plan, including the route plan, hunt pilots transformations, and URI dialing.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Call Admission Control Configuration, on page 203</td>
<td>Complete the configurations in this part of the guide to configure call admission control.</td>
</tr>
<tr>
<td>Step 5</td>
<td>End User Configuration, on page 223</td>
<td>Complete the configurations in this part of the guide to configure roles and access control groups, and to configure end users and their profiles.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Endpoint Device Configuration, on page 281</td>
<td>Complete the configurations in this part of the guide to configure endpoint devices such as analog telephone adaptors (ATAs), Cisco IP Phones, third-party SIP phones, and software-based endpoints. This part also describes how to configure templates and profiles for endpoint devices, and how to enable diagnostics and reporting for Cisco IP phones.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Integrate Applications, on page 385</td>
<td>Complete the configurations in this part of the guide to configure Cisco Unified Communications Manager to work with other applications, such as CTI applications, Cisco Unity Connection, Cisco IM and Presence Service, Unified Contact Center Enterprise and Express, and Cisco TelePresence.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Media Resources Configuration Task Flow, on page 435</td>
<td>Complete the configurations in this part of the guide to configure audio and video resources, annunciators, transcoders, media termination points, and conference bridges.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Registering Devices, on page 503</td>
<td>Complete the configurations in this part of the guide to automatically register endpoint devices, or to manually register endpoint devices.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong> Advanced Call Handling Configuration, on page 537</td>
<td>Complete the configurations in this part of the guide to configure call controls, including how calls are advertised, queued, and throttled. This part also describes how to configure logical partitioning, calling party normalization, geolocation conveyance, automated alternate routing (AAR), and multilevel precedence and preemption (MLPP).</td>
<td></td>
</tr>
<tr>
<td><strong>Step 11</strong> Cisco Unified Communications Manager TCP and UDP Port Usage Overview, on page 653 IM and Presence Service Port Usage Overview, on page 671</td>
<td>This part of the guide provides information about the ports that Unified Communications Manager and the IM and Presence Service use for intracluster connections and for communication with external applications or devices.</td>
<td></td>
</tr>
</tbody>
</table>
PART I

Configure Initial Parameters for the System

• Initial Configuration Overview, on page 9
• Configure System Licensing, on page 11
• Configure Server Information, on page 19
• Configure System and Enterprise Parameters, on page 25
• Configure Service Parameters, on page 35
• Configure Core Settings for Device Pools, on page 41
CHAPTER 2

Initial Configuration Overview

- About Initial Configuration, on page 9
- Initial Configuration Task Flow, on page 9

About Initial Configuration

The chapters in this section describe the initial setup tasks that you must complete before you begin to configure the call control system.

Initial Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>System Licensing Configuration Task Flow, on page 12</td>
<td>Manage the license requirements for your system.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Server Configuration Task Flow, on page 19</td>
<td>Configure basic server information, such as the server name and port settings.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Initial System and Enterprise Configuration Task Flow, on page 25</td>
<td>Configure the system-wide parameters that are required when you set up a node for the first time.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Service Parameters Configuration Task Flow, on page 35</td>
<td>Configure the service parameters that are required when you set up a node for the first time.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Core Settings for Device Pools Configuration Task Flow, on page 47</td>
<td>Configure core system settings, such as a server group, time zone information, and a region (codec selection). These settings are fundamental and become the foundation for basic device pools.</td>
</tr>
</tbody>
</table>
CHAPTER 3

Configure System Licensing

- System Licensing Overview, on page 11
- System Licensing Prerequisites, on page 12
- System Licensing Configuration Task Flow, on page 12
- Licensing Interactions and Restrictions, on page 16

System Licensing Overview

Your system uses centralized license management, whereby license usage moves away from the device-based licensing and towards user-based licensing. This user-centered model matches the overall requirements of your system and what your users actually use—for example, multiple devices that are associated with one user. From this model, licensing for your system is determined by the total number of users itself, user features, and configured devices.

The following licensing options are available to cover your users' needs:

**Cisco Unified Workspace Licensing**

Cisco Unified Workspace Licensing (UWL) provides the most popular bundles of Cisco Collaboration applications and services in a cost-effective, simple package. It includes soft clients, applications server software, and licensing on a per-user basis.

**Cisco User Connect Licensing**

A per-user based license for individual Cisco Unified Communications applications, that includes the applications server software, user licensing, and a soft client. Depending the type of device and number of devices that you require, User Connect Licensing (UCL) is available in essential, basic, enhanced, and enhanced plus versions.

For more information about these license types and the versions in which they are available, see http://www.cisco.com/c/en/us/products/unified-communications/unified-communications-licensing/index.html.

**Cisco Prime License Manager**

Cisco Prime License Manager consolidates all license requirements from all Cisco Unified Communications Managers in your system and compares the total requirements to the number of available licenses installed. When you provision users, phones, or other services in your system, license requirements are calculated. After this step, the Cisco Unified Communications Manager sends the corresponding license requirements to Cisco
Prime License Manager. The license then compares your system license requirements with the available licenses installed, and reports back license compliance or non-compliance. They continually synchronize data and keep your system licensing snapshot up to date.

**System Licensing Prerequisites**


If you are migrating your licenses from a previous release, verify the following requirements:


- Apply all pre-9.x licenses to your Unified Communications Manager before you upgrade.


To obtain migration support, select one of the following options:

- Open your service request through the web: [https://tools.cisco.com/ServiceRequestTool/scm/mgmt/case](https://tools.cisco.com/ServiceRequestTool/scm/mgmt/case)
- Open any service request through licensing@cisco.com (include Cisco.com user ID)

### System Licensing Configuration Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Integrate Your Cluster with Cisco Prime License Manager, on page 13</td>
<td>Add your Unified Communications Manager publisher node to the Cisco Prime License Manager as a product instance. This task ensures synchronization of licensing requirements and usage between the two systems. Cisco Prime License Manager polls your system every 24 hours. You add the cluster by adding the publisher node.</td>
</tr>
</tbody>
</table>
Integrate Your Cluster with Cisco Prime License Manager

Add your Unified Communications Manager publisher node to the Cisco Prime License Manager as a product instance. This task ensures synchronization of licensing requirements and usage between the two systems. Cisco Prime License Manager polls your system every 24 hours. You add the cluster by adding the publisher node.

Before you begin

Install the remaining licenses to the product instance that you want to add. This steps ensure that those licenses are eligible for migration.

Check the status of you account by using the following command: show accountlocking. You must set the account locking setting to disabled to avoid a 401 error when you attempt to add the product instance.

Procedure

Step 1 Log in to Cisco Prime License Manager using the application username and password that you created when you completed the installation.

Step 2 Select Product Instances.

Step 3 Click Add. The Product Add dialog box appears.

Step 4 Enter the following information:
  - Name
  - Description (optional)
  - Product Type
  - Hostname/IP Address
  - Username
  - Password

Note Credentials are the OS Administration username and password of the product.

Step 5 Click OK to add the product instance.
**Step 6**

Once the product instance has been successfully added, the product appears in the Product Instances table.

*Note*  
On the Product Instances page, click **Synchronize Now** to request the licensing information from the new product. If you do not synchronize, current product instance information will not appear in Cisco Prime License Manager until the next scheduled synchronization is completed.

---

**Synchronize Your Cluster**

Synchronize your cluster with Cisco Prime License Manager in order to upload all device data from the cluster and allow the license manager to allocate the licenses.

**Before you begin**

*Note*  
If you have more than one cluster, we recommend that you add all your clusters before you perform your first synchronization.

**Procedure**

**Step 1**  
In Cisco Prime License Manager, select **Product Instances**.

**Step 2**  
Click **Synchronize Now**.

A message appears that confirms the synchronization is complete.

---

**Create a License Plan**

To obtain new licenses for your system, walk through the add licenses wizard to prepare your license request.

**Before you begin**

*Synchronize Your Cluster*, on page 14

**Procedure**

**Step 1**  
From the **Licenses > Planning** window in Cisco Prime License Manager, click **Create an Add Licenses Plan**.

**Step 2**  
From the **Choose Product** section, select the product type and license version of the product to which you will be adding a licence. Click Next.

**Step 3**  
From the **License Counts** section, adjust the number of licenses that will be allocated to each type of license and click Save to save your changes for that license type. You may also choose to run a compliance check by clicking **Run Compliance Check**, or reset the license values by clicking **Reset Values**. After the number of licenses has been set, click Next.
Clicking the arrow next to each license type reveals additional information about that license type.

**Step 4**  
If your compliance check fails, you can return to **License Counts** to make additional changes. If the compliance check passes, click **Next** to move to the next section.

**Step 5**  
In **Summary and Next Steps** section, you can view and save a summary of the changes you made. You can also enter your own summary name and description.

**Step 6**  
To view the summary, click **View Summary**. The **Save Summary in Cisco Prime License Manager** option is selected by default. A default name for the summary also appears in the **Name** field using the format <product-type>-add-<date-time-stamp> format. Instructions for placing your order and fulfilling your licenses also appear in this section.

**Step 7**  
Click **Finish**

---

**What to do next**

*Install the License File, on page 15*

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**Install the License File**

Follow the license fulfillment procedure to install the applicable licenses for your system. These steps ensure that your system is in compliance.

**Procedure**

**Step 1**  
From the Cisco Prime License Manager main menu, select **Licenses > Fulfillment**.

**Step 2**  
In eFulfillment mode, click **Fulfill Licenses from PAK**.

**Step 3**  
Select **Add licenses from a new PAK** and enter the Product Authorization Key (PAK) code.

If you have previously entered PAKs in Cisco Prime License Manager, you can select **Add licenses from an already-installed PAK that supports partial fulfillment**. After you have selected that option, select the existing PAK code from the drop-down menu.

**Step 4**  
Click **Next**. If prompted for your Cisco.com account information, enter the username and password you entered when you registered at Cisco.com.

**Step 5**  
Click **OK**.

If there are licenses remaining to be fulfilled (and the PAK username and password are validated), the **Fulfill Licenses** section appears.

Licenses can only be fulfilled using the cisco.com account to which they were initially issued.

**Step 6**  
The licenses within the PAK are listed by SKU name. The numbers of each license are categorized under a number of headings to indicate how many have been fulfilled and how many are remaining.

You can specify the number of licenses you want to fulfill by selecting **Fulfill** in the Actions column for that license type. In the **Fulfill Licenses** window you can specify the license version, feature, or both, and click **Save** then click **OK** to close the window. The updated count now appears in the Fulfill column of the Fulfill Licenses table.
Some PAKs are not eligible for partial fulfillment. These PAKs are packaged together, so they can only be fulfilled at a single Cisco Prime License Manager in a single transaction. For example, an NFR (not-for-resale) order is sold as a package with 20 CUWL Pro Unified CM and Unity Connection licenses and five TelePresence Room licenses.

If licenses are listed as "Fulfilled" (under "Before Fulfillment" in the Fulfill Licenses table), those licenses have previously been fulfilled by this or another Cisco Prime License Manager.

Once you have fulfilled your licenses as selected, you may wish to click **Run Compliance Check** to ensure that you are in compliance.

**Step 7**
Click **Next** to review your changes. If you are not satisfied with your changes, click **Previous** to return to the Fulfill Licenses section. If you are satisfied with the changes, click **Next** to move to the next section.

**Step 8**
Click **Next** in the Fulfill Licenses section opens the Transaction Options and License Agreement section. In this section, you may enter a description (optional). You may also associate this transaction with a saved license summary by selecting that option and then selecting the name of the license summary from the drop-down list.

**Step 9**
Select the checkbox to accept the conditions of the End User License Agreement.

**Step 10**
Click **Finish**.
Upon successful completion of the e-Fulfillment process, the new fulfillment appears in the License Fulfillment table.

---

### Licensing Interactions and Restrictions

- **System Licensing Interactions**, on page 16
- **System Licensing Restrictions**, on page 17

### System Licensing Interactions

**Table 1: System Licensing Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extension Mobility</td>
<td>Users who are configured with extension mobility but have no associated devices do not require an essential license.</td>
</tr>
<tr>
<td>MGCP</td>
<td>MGCP FXS ports do not require licenses, because they are not considered to be analog phones.</td>
</tr>
</tbody>
</table>
Multi-purpose and immersive TelePresence devices are licensed under a separate device license type of the TelePresence Room license. The TelePresence Room license covers both the TelePresence device and registered phone, only if the same user ID is entered as the OwnerUserID field for the TelePresence device as is entered for the phone.

If the same user ID is not entered as the owner user ID for both the TelePresence device and the phone, then the device and phone are not associated and two licenses are required: one TelePresence Room license for the device and one enhanced user connect license (UCL) for the phone. The TelePresence touch device does not register, and therefore does not require a separate license or the owner user ID association.

### System Licensing Restrictions

**Table 2: System Licensing Restrictions**

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Number of Devices Per User</td>
<td>The essential, basic, and enhanced User Connect Licensing (UCL) licenses support users with one associated device, where the user ID is entered in the OwnerUserId field of one device. The Enhanced Plus UCL license supports users with two associated devices. The Unified Workspace Licensing (UWL) Standard and UWL premium support users with three and up to ten associated devices.</td>
</tr>
<tr>
<td>Product Authorization Keys</td>
<td>Some product authorization keys (PAKs) are not eligible for partial fulfillment. These PAKs are packaged together, so they can only be fulfilled at a single Cisco Prime License Manager. For example, a not-for-resale (NFR) order is sold as a package with 20 CUWL Pro Unified Communications Manager and Unity Connection licenses as well as five TelePresence Room licenses.</td>
</tr>
</tbody>
</table>
CHAPTER 4

Configure Server Information

- System Information Overview, on page 19
- Server Configuration Task Flow, on page 19

System Information Overview

This chapter describes how to configure the properties of the Unified Communications Manager node.

Note

All Unified Communications products such as Unified Communications Manager, Cisco Unity Connections, and Cisco IM and Presence, and so on, have only one interface. Thus, you can assign only one IP address for each of these products.

Server Configuration Task Flow

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure Server Information, on page 20</td>
<td>Specify a name for the Unified Communications Manager node, and add a description.</td>
</tr>
<tr>
<td>Step 2 Configure Ports , on page 20</td>
<td>Configure the following ports:</td>
</tr>
<tr>
<td></td>
<td>• Ethernet Phone Port</td>
</tr>
<tr>
<td></td>
<td>• MGCP Listen Port</td>
</tr>
<tr>
<td></td>
<td>• MGCP Keep-alive Port</td>
</tr>
<tr>
<td></td>
<td>• SIP Phone Port</td>
</tr>
<tr>
<td></td>
<td>• SIP Phone Secure Port</td>
</tr>
</tbody>
</table>
Configure Server Information

Specify a name for the Unified Communications Manager node, and add a description. You can also use this procedure to view the following read-only information:

- The computer telephony integration identification (CTI ID).
- The server where this Unified Communications Manager is installed.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified Communications Manager Administration, select <strong>System &gt; Cisco Unified CM</strong>. The <strong>Find and List Cisco Unified CMs</strong> window appears.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Enter the appropriate search criteria and click <strong>Find</strong>. All matching Cisco Unified Communications Managers are displayed.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Select the <strong>Cisco Unified CM</strong> that you want to view. The <strong>Cisco Unified CM Configuration</strong> window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Name</strong> field, enter the name that you want to assign to this Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the <strong>Description</strong> field, enter a description for the node. The description can include up to 50 characters in any language, but it cannot include double-quotes (&quot;), percentage sign (%), ampersand (&amp;), back-slash (), or angle brackets (&lt;&gt;).</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

Configure Ports

Use this procedure to change the port settings used for connections such as SCCP device registration, SIP device registration, and MGCP gateway connections.

**Note**

Normally, you need not change the default port settings. Use this procedure only if you really want to change the defaults.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
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</thead>
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<tr>
<td>Step 4</td>
<td>Navigate to the <strong>Cisco Unified Communications Manager TCP Port Settings for this Server</strong> section.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>
Step 6  Click **Apply Config**.
Step 7  Click **OK**.

### Port Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ethernet Phone Port</td>
<td>The system uses this TCP port to communicate with the Cisco Unified IP Phones (SCCP only) on the network.</td>
</tr>
<tr>
<td></td>
<td>• Accept the default port value of 2000 unless this port is already in use on your system. Choosing 2000 identifies this port as non-secure.</td>
</tr>
<tr>
<td></td>
<td>• Ensure all port entries are unique.</td>
</tr>
<tr>
<td></td>
<td>• Valid port numbers range from 1024 to 49151.</td>
</tr>
<tr>
<td>MGCP Listen Port</td>
<td>The system uses this TCP port to detect messages from its associated MGCP gateway.</td>
</tr>
<tr>
<td></td>
<td>• Accept the default port of 2427 unless this port is already in use on your system.</td>
</tr>
<tr>
<td></td>
<td>• Ensure all port entries are unique.</td>
</tr>
<tr>
<td></td>
<td>• Valid port numbers range from 1024 to 49151.</td>
</tr>
<tr>
<td>MGCP Keep-alive Port</td>
<td>The system uses this TCP port to exchange keepalive messages with its associated MGCP gateway.</td>
</tr>
<tr>
<td></td>
<td>• Accept the default port of 2428 unless this port is already in use on your system.</td>
</tr>
<tr>
<td></td>
<td>• Ensure all port entries are unique.</td>
</tr>
<tr>
<td></td>
<td>• Valid port numbers range from 1024 to 49151.</td>
</tr>
<tr>
<td>SIP Phone Port</td>
<td>This field specifies the port number that Unified Communications Manager uses to listen for SIP line registrations over TCP and UDP.</td>
</tr>
<tr>
<td>SIP Phone Secure Port</td>
<td>This field specifies the port number that the system uses to listen for SIP line registrations over TLS.</td>
</tr>
</tbody>
</table>

### Hostname Configuration

The following table lists the locations where you can configure a host name for the Unified Communications Manager server, the allowed number of characters for the host name, and the recommended first and last characters for the host name. Be aware that, if you do not configure the host name correctly, some components in Unified Communications Manager, such as the operating system, database, installation, and so on, may not work as expected.
Table 3: Host Name Configuration in Cisco Unified Communications Manager

<table>
<thead>
<tr>
<th>Host Name Location</th>
<th>Allowed Configuration</th>
<th>Allowed Number of Characters</th>
<th>Recommended First Character for Host Name</th>
<th>Recommended Last Character for Host Name</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Host Name/IP Address field</strong>&lt;br&gt;System &gt; Server in Cisco Unified Communications Manager Administration</td>
<td>You can add or change the host name for a server in the cluster.</td>
<td>2-63</td>
<td>alphabetic</td>
<td>alphanumerical</td>
</tr>
<tr>
<td><strong>Hostname field</strong>&lt;br&gt;Cisco Unified Communications Manager installation wizard</td>
<td>You can add the host name for a server in the cluster.</td>
<td>1-63</td>
<td>alphabetic</td>
<td>alphanumerical</td>
</tr>
<tr>
<td><strong>Hostname field</strong>&lt;br&gt;Settings &gt; IP &gt; Ethernet in Cisco Unified Communications Operating System</td>
<td>You can change, not add, the host name for a server in the cluster.</td>
<td>1-63</td>
<td>alphabetic</td>
<td>alphanumerical</td>
</tr>
<tr>
<td><strong>set network hostname</strong>&lt;br&gt;hostname&lt;br&gt;Command Line Interface</td>
<td>You can change, not add, the host name for a server in the cluster.</td>
<td>1-63</td>
<td>alphabetic</td>
<td>alphanumerical</td>
</tr>
</tbody>
</table>

Tip: The host name must follow the rules for ARPANET host names. Between the first and last character of the host name, you can enter alphanumeric characters and hyphens.

Before you configure the host name in any location, review the following information:

- The Host Name/IP Address field in the Server Configuration window, which supports device-to-server, application-to-server, and server-to-server communication, allows you to enter an IPv4 address in dotted decimal format or a host name.

After you install the Unified Communications Manager publisher node, the host name for the publisher automatically displays in this field. Before you install a Unified Communications Manager subscriber node, enter either the IP address or the host name for the subscriber node in this field on the Unified Communications Manager publisher node.

In this field, configure a host name only if Unified Communications Manager can access the DNS server to resolve host names to IP addresses; make sure that you configure the Cisco Unified Communications Manager name and address information on the DNS server.

Tip: In addition to configuring Unified Communications Manager information on the DNS server, you enter DNS information during the Cisco Unified Communications Manager installation.

- During the installation of the Unified Communications Manager publisher node, you enter the host name, which is mandatory, and IP address of the publisher node to configure network information; that is, if you want to use static networking.
During the installation of a Unified Communications Manager subscriber node, you enter the hostname and IP address of the Unified Communications Manager publisher node, so that Unified Communications Manager can verify network connectivity and publisher-subscriber validation. Additionally, you must enter the host name and the IP address for the subscriber node. When the Unified Communications Manager installation prompts you for the host name of the subscriber server, enter the value that displays in the Server Configuration window in Cisco Unified Communications Manager Administration; that is, if you configured a host name for the subscriber server in the Host Name/IP Address field.
Hostname Configuration
Initial System and Enterprise Parameters Overview

Consider the following system-wide parameters when you set up a Unified Communications Manager node for the first time. You can modify system-wide parameters for your deployment if needed; however, the recommended default settings should work in most cases.

• Set the fall-back connection monitor duration for IP phones.
• Allow searches of the corporate directory for all users.
• Set the Fully Qualified Directory Number (FQDN) for the cluster and the top-level domain for the organization.
• Set the Cisco Jabber start condition for video.
• (Optional) Enable Multi-Level Precedence and Preemption (MLPP) if your cluster uses MLPP.
• (Optional) Enable IPv6 if your network uses IPv6.
• (Optional) Enter a remote syslog server name.
• (Optional) Set up call trace log to troubleshoot your deployment.
• (Optional) Enable dependency records.

Initial System and Enterprise Configuration Task Flow

Before you begin

Set up your Unified Communications Manager node and port settings.
### Configure Initial Parameters for the System

#### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure the system-wide parameters that are required for an initial setup of your Unified Communications Manager node. For a list of the recommended system settings, see Common Enterprise Parameters, on page 28.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure the enterprise parameter that is required to allow Cisco Jabber to perform certificate-based authentication with the IdP in a controlled mobile device management (MDM) deployment.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure the enterprise parameter through Unified Communications Manager to enable SAML SSO for Real-Time Monitoring Tool (RTMT).</td>
</tr>
</tbody>
</table>

#### What to do next

Configure some core settings for device pools to lay a foundation of common settings to apply across all devices that are configured in the Unified Communications Manager cluster, see Core Settings for Device Pools Configuration Task Flow, on page 47.

### Configure Initial System and Enterprise Parameters

You can use Cisco Unified Communications Manager Administration to configure system and enterprise parameters for your particular deployment. Although we've listed parameters that are important for an initial system setup, the recommended default settings work for most deployments.

Parameters that are useful for troubleshooting, such as enabling call trace logs, should be disabled after you are finished troubleshooting so that network performance is not impacted.

Most parameters require that you reset all devices for the changes to take effect. Consider completing all configuration steps before you perform a reset of all devices. We recommend that you reset all devices during off-peak hours.

#### Note

From Release 10.0(1), the same enterprise parameters are used on Unified Communications Manager and IM and Presence Service. If you change the value of an enterprise parameter on IM and Presence Service, the changed value is automatically updated to Unified Communications Manager.

#### Procedure

**Step 1**

In Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.
Configure Initial Parameters for the System

Configure Initial System and Enterprise Parameters

Step 2 In the **Enterprise Parameters Configuration** section, enter the number of seconds in the **Connection Monitor Duration** field before a IP phone in the cluster falls back to the primary node when the TCP connection becomes available, then click **Save**. The default value is 120 seconds.

**Tip** To apply the changes to all affected devices in the cluster without resetting all devices, click **Apply Config**, and then click **OK**.

Step 3 In the **User Data Service Parameters** section, select **True** in the **Enable All User Search** field to allow users to search the corporate directory for all users when no last name, first name, or directory number is specified.

Step 4 In the **Clusterwide Domain Configuration** section, set up the clusterwide domain.

a) Enter the top-level domain for the organization in the **Organization Top Level Domain** field. The maximum length is 255 characters.

b) Enter the Fully Qualified Domain Name (FQDN) for the cluster in the **Cluster Fully Qualified Domain Name** field. The maximum length is 255 characters.

   Multiple FQDNs must be separated by a space. Wildcards can be specified within an FQDN using an asterisk (*). Example: cluster-1.cisco.com *.cisco.com.

Step 5 In the Cisco Jabber section, select **False** in the **Never Start Call with Video** field.

Step 6 (Optional) In the **MLPP and Confidential Access Level Parameters** section, enter the Multi-Level Precedence and Preemption (MLPP) domain and enable devices to use MLPP.

a) Enter the domain for the MLPP service in the **MLPP Domain Identifier** field. This parameter accepts hexadecimal values starting with 0x.

b) Select **MLPP indication turned on** in the **MLPP Indication Status** field.

Step 7 (Optional) In the **IPv6** section, set the **Enable IPv6** field to **True**.

Step 8 (Optional) In the **Cisco Syslog Agent** section, enter the name or IP address of the remote Syslog server in the **Remote Syslog Server Name 1** field. If a server name is not specified, Cisco Unified Serviceability does not send the Syslog messages.

Step 9 (Optional) In the **Call Trace Log Configuration for Session Trace** section, set up the call trace log to allow the collection of SIP call information for session traces.

   The Session Trace feature in the Real-Time Monitoring Tool (RTMT) uses this information to generate call flow diagrams that are useful for troubleshooting.

   a) Set the **Enable Call Trace Log** field to **True**.

   b) Enter the maximum number of SIP call trace log files that Unified Communications Manager can generate in the **Max Number of Call Trace Log Files** field.

      The default value is 2000. Valid range is from 5 to 4000.

   c) Enter the maximum file size, in megabytes, of the SIP call trace log files in the **Call Trace Log** field.

      The default value is 2. Valid range is from 1 to 10.

   **Note** Some performance degradation can occur during periods of high SIP call traffic. To reduce the impact on system performance, set the Cisco CallManager service parameter called **Log Call-Related REFER/NOTIFY/SUBSCRIBE SIP Messages for Session Trace** to False. This will omit the REFER, NOTIFY, and SUBSCRIBE messages from the SIP call tracing.

Step 10 In the in the **CCMAadmin Parameters** section, select **True** in the **Enable Dependency Records** field.

Step 11 Click **Save**.
Step 12  Click **Reset**, and then click **OK** to reset all devices.

We recommend that you reset all devices during off-peak hours.

**Tip**  To reset all devices, you can reset every device pool in the system.

---

**Related Topics**

- [Common Enterprise Parameters](#), on page 28

---

### Common Enterprise Parameters

The following table lists common enterprise parameters that are used to set enterprise settings such as Organization Top-Level Domain or Cluster Fully Qualified Domain Name. For a detailed list, use the **System > Enterprise Parameters** menu in Cisco Unified CM Administration.

#### Table 4: Common Enterprise Parameters for an Initial Unified Communications Manager Setup

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Enterprise Parameters</strong></td>
<td></td>
</tr>
</tbody>
</table>
| Connection Monitor Duration | If an IP phone in the cluster registers on a secondary node, use this parameter to set the amount of time that the IP phone waits before it falls back and re-registers with the primary node after the primary node becomes available. This parameter affects all secure devices for a specific Secure Survivable Remote Site Telephony (SRST) router.  
  For more information, see *Security Guide for Cisco Unified Communications Manager*.  
  Default: 120 seconds  
  Restart all services for the changes to take effect. |
| **CCMAadmin Parameters** |                                                                                                                                              |
| Enable Dependency Records | This parameter is used to display dependency records that are required for troubleshooting. Displaying the dependency records may be beneficial during an initial system setup.  
  Displaying the dependency records could lead to high CPU usage spikes and could impact call processing. To avoid possible performance issues, disable this parameter after the system setup is complete. We recommend displaying dependency records only during off-peak hours or during a maintenance window.  
  When enabled, you can select **Dependency Records** from the **Related Links** drop-down list, which is accessible from most configuration windows using Unified Communications Manager.  
  Default: False |

---

**User Data Service Parameters**
### Common Enterprise Parameters

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Description</th>
</tr>
</thead>
</table>
| Enable All User Search         | This parameter allows you to search the corporate directory for all users when no last name, first name, or directory number is specified. This parameter also applies to directory searches on the **Cisco CallManager Self Care** (CCMUser) window.  
  Default: True                                                                                                                      |

#### Clusterwide Domain Configuration

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Description</th>
</tr>
</thead>
</table>
| Organization Top Level Domain   | This parameter defines the top-level domain for the organization. For example, cisco.com.                                                                                                                    
  Maximum length: 255 characters  
  Allowed values: A valid domain using upper and lowercase letters, numbers (0-9), hyphens, and dots (as a domain label separator). Domain labels must not start with a hyphen. The last label must not start with a number. For example, this domain is invalid -cisco.1om. |
| Cluster Fully Qualified Domain Name | This parameter defines one or more Fully Qualified Domain Names (FQDN) for the cluster. Multiple FQDNs must be separated by a space. Specify wildcards within an FQDN using an asterisk (*). Example: cluster-1.cisco.com  
  Requests containing URLs, such as SIP calls, that have a host portion that matches any of the FQDNs in this parameter are routed to that cluster and the attached devices.  
  Maximum length: 255 characters  
  Allowed values: An FQDN or a partial FQDN using the * wildcard. Upper and lowercase letters, numbers (0-9), hyphens, and dots (as a domain label separator). Domain labels must not start with a hyphen. The last label must not start with a number. For example, this domain is invalid -cisco.1om. |

#### IPv6
<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable IPv6</td>
<td>This parameter determines whether Unified Communications Manager can negotiate Internet Protocol Version 6 (IPv6) and whether phones are allowed to advertise IPv6 capability. IPv6 must be enabled on all other network components including on the platform of all nodes before you enable this parameter. Otherwise, the system continues to run in IPv4-only mode. This is a required field. Default: False (IPv6 is disabled) You must restart the following services for the IPv6 parameter change to take effect, and the affected services in the Instant Messaging and Presence cluster.</td>
</tr>
<tr>
<td></td>
<td>• Cisco CallManager</td>
</tr>
<tr>
<td></td>
<td>• Cisco IP Voice Media Streaming App</td>
</tr>
<tr>
<td></td>
<td>• Cisco CTIManager</td>
</tr>
<tr>
<td></td>
<td>• Cisco Certificate Authority Proxy Function</td>
</tr>
</tbody>
</table>

**Cisco Syslog Agent**

| Remote Syslog Server Name 1 | Enter the name or IP address of the remote Syslog server. Cisco Unified Serviceability do not send the Syslog messages if a server name is not specified. This parameter is required only if you are using the Syslog server for logs. Maximum length: 255 characters Allowed values: A valid remote Syslog server name using upper and lowercase letters, numbers (0-9), hyphens, and dots. Do not specify another Unified Communications Manager node as the destination. |

**Cisco Jabber**

| Never Start Call with Video | This parameter determines if video is sent when a video call starts. Select True to start video calls without immediately sending video. Anytime during the video call, you can choose to start sending your video. This parameter overrides any Instant Messaging and Presence preferences. When set to False, video calls start according to the preferences set in Instant Messaging and Presence. Default: False. |

**SSO and OAuth Configuration**
### Parameter Name: SSO Login Behavior for iOS

**Description:**

This parameter is required to allow Cisco Jabber to perform the certificate-based authentication with the IdP in a controlled mobile device management (MDM) deployment.

The **SSO Login Behavior for iOS** parameter includes the following options:

- **Use Embedded Browser**—If you enable this option, Cisco Jabber uses the embedded browser for the SSO authentication. Use this option to allow iOS devices prior to version 9 to use SSO without cross-launching into the native Apple Safari browser.

- **Use Native Browser**—If you enable this option, Cisco Jabber uses the Apple Safari framework on an iOS device to perform the certificate-based authentication with an Identity Provider (IdP) in the MDM deployment.

**Note:** We do not recommend configuring this option, except in a controlled MDM deployment, because using a native browser is not as secure as the using the embedded browser.

This is a required field.

Default: Use the embedded browser (WebView).

---

### Parameter Name: OAuth with Refresh Login Flow

**Description:**

This parameter controls the login flow used by clients such as Cisco Jabber when connecting to Unified Communication Managers.

- **Enabled**—If you enable this option, clients can use an OAuth-based Fast Login flow to provide a quicker and streamlined login experience, without requiring the user input to re-log in. For example, due to a network change. The option requires support from the other components of the Unified Communications solution, such as Expressway and Unity Connection (compatible versions with the refresh login flow enabled).

- **Disabled**—If you enable this option, the existing behavior is preserved and is compatible with older versions of other system components.

**Note:**

For Mobile and Remote Access deployment with Cisco Jabber, we recommend enabling this parameter only with a compatible version of Expressway that supports OAuth with Refresh login flow. Incompatible version may impact the Cisco Jabber functionality. Please refer the specific product documents for supported version and configuration requirements.

This is a required field.

Default: Disabled.
This parameter is configured to enable SAML SSO for Real-Time Monitoring Tool (RTMT).

The **Use SSO for RTMT** parameter includes the following options:

- **True**—If you choose this option, RTMT displays the SAML SSO-based IdP sign-in window.

  **Note** When you perform a fresh install, the default value of the **Use SSO for RTMT** parameter appears as **True**.

- **False**—If you choose this option, RTMT displays the basic authentication sign-in window.

  **Note** When you perform an upgrade from a Cisco Unified Communications Manager version where **Use SSO for RTMT** parameter does not exist, the default value of this parameter in the newer version appears as **False**.

This is a required field.

Default: True.

### Configure SSO Login Behavior for Cisco Jabber on iOS

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **System > Enterprise Parameters**.

**Step 2**  
To configure the opt-in control, in the SSO Configuration section, choose the **Use Native Browser** option for the **SSO Login Behavior for iOS** parameter:

**Note**  
The **SSO Login Behavior for iOS** parameter includes the following options:

- **Use Embedded Browser**—If you enable this option, Cisco Jabber uses the embedded browser for SSO authentication. Use this option to allow iOS devices prior to version 9 to use SSO without cross-launching into the native Apple Safari browser. This option is enabled by default.

- **Use Native Browser**—If you enable this option, Cisco Jabber uses the Apple Safari framework on an iOS device to perform certificate-based authentication with an Identity Provider (IdP) in the MDM deployment.

  **Note** We don't recommend to configure this option, except in a controlled MDM deployment, because using a native browser is not as secure as the using the embedded browser.

**Step 3**  
Click **Save**.
## Configure SSO for RTMT

### Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>System &gt; Enterprise Parameters</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>To configure SSO for RTMT, in the SSO Configuration section, choose <strong>True</strong> for the <strong>Use SSO for RTMT</strong> parameter:</td>
</tr>
</tbody>
</table>

**Note** The **Use SSO for RTMT** parameter includes the following options:

- **True**—If you choose this option, RTMT displays the SAML SSO-based IdP sign-in window.
  
  **Note** When you perform a fresh install, the default value of the **Use SSO for RTMT** parameter appears as **True**.

- **False**—If you choose this option, RTMT displays the basic authentication sign-in window.
  
  **Note** When you perform an upgrade from a Cisco Unified Communications Manager version where **Use SSO for RTMT** parameter does not exist, the default value of this parameter in the newer version appears as **False**.

| Step 3 | Click **Save**. |
Configure Initial Parameters for the System

Configure SSO for RTMT
Configure Service Parameters

- Service Parameters Overview, on page 35
- Service Parameters Configuration Task Flow, on page 35

Service Parameters Overview

Service parameters let you configure different services on selected Unified Communications Manager servers. Unlike enterprise parameters, which apply to all services, each service gets configured with a separate set of service parameters.

Service parameters let you configure settings for the following two types of services, both of which can be activated within Cisco Unified Serviceability:

- **Feature Services** - These services are used to run certain system features. You must turn feature services on in order to use them.

- **Network Services** - Network services are on by default, but you can stop and start (or restart) a network service for troubleshooting purposes. These services include services that allow system components like the database and platform to function properly.

You can view service parameter field descriptions for service parameters by by clicking the ? icon within the Service Parameter Configuration window, or by clicking on one of the parameter names.

---

**Note**
If you deactivate a service, Unified Communications Manager retains any updated service parameter values. If you start the service again, Unified Communications Manager sets the service parameters to the changed values.

---

Service Parameters Configuration Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Activate Essential Services, on page 36</td>
<td>You can activate and deactivate services on the node using Cisco Unified Serviceability. For a</td>
</tr>
</tbody>
</table>
Activate Essential Services

Use this procedure to activate services across the cluster.

For a list of recommended services for publisher nodes and subscriber nodes, see the following topics:

- Recommended Services for Publisher Nodes, on page 36
- Recommended Services for Subscriber Nodes, on page 38

Procedure

**Step 1**  
From Cisco Unified Serviceability, choose Tools > Service Activation.

**Step 2**  
Select a Server from the drop-down menu and click Go.  
The services and their current status display.

**Step 3**  
Activate and deactivate the services that you want:  
- To activate a service, check the check box beside the service that you want to activate.
- To deactivate a service, uncheck the check box beside the service that you want to deactivate.

**Step 4**  
Click Save.  
Service activation may take a few minutes to complete. refresh the page to confirm the status change.

Recommended Services for Publisher Nodes

The following table lists recommended services for a Unified Communications Manager publisher node when using a non-dedicated TFTP server.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>list of the recommended services for publisher nodes, see Recommended Services for Publisher Nodes, on page 36. For a list of the recommended services for subscriber nodes, see Recommended Services for Subscriber Nodes, on page 38.</td>
<td>Configure service parameters for the Cisco Unified Communications Manager publisher node and for subscriber nodes in the cluster.</td>
</tr>
<tr>
<td>Configure Service Parameters, on page 38</td>
<td>You can display the services for your nodes using Cisco Unified Communications Manager Administration and Cisco Unified Serviceability. To view service parameter settings and parameter descriptions, use Cisco Unified Communications Manager Administration.</td>
</tr>
<tr>
<td>View Clusterwide Service Parameter Settings , on page 39</td>
<td></td>
</tr>
</tbody>
</table>

---

**Activate Essential Services**

Use this procedure to activate services across the cluster.

For a list of recommended services for publisher nodes and subscriber nodes, see the following topics:

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**Recommended Services for Publisher Nodes**

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<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>list of the recommended services for publisher nodes, see Recommended Services for Publisher Nodes, on page 36. For a list of the recommended services for subscriber nodes, see Recommended Services for Subscriber Nodes, on page 38.</td>
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</tr>
<tr>
<td>Configure Service Parameters, on page 38</td>
<td>You can display the services for your nodes using Cisco Unified Communications Manager Administration and Cisco Unified Serviceability. To view service parameter settings and parameter descriptions, use Cisco Unified Communications Manager Administration.</td>
</tr>
<tr>
<td>View Clusterwide Service Parameter Settings , on page 39</td>
<td></td>
</tr>
</tbody>
</table>
## Table 5: Recommended Publisher Node Services for Non-Dedicated TFTP Server Deployments

<table>
<thead>
<tr>
<th>Type</th>
<th>Service Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>CM Services</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td></td>
<td>Cisco Unified Mobile Voice Access Services</td>
</tr>
<tr>
<td></td>
<td>Cisco IP Voice Media Streaming App</td>
</tr>
<tr>
<td></td>
<td>Cisco CTIManager</td>
</tr>
<tr>
<td></td>
<td>Cisco Extended Functions</td>
</tr>
<tr>
<td></td>
<td>Cisco Intercluster Lookup Service</td>
</tr>
<tr>
<td></td>
<td>Cisco Location Bandwidth Manager</td>
</tr>
<tr>
<td></td>
<td>Cisco TFTP</td>
</tr>
<tr>
<td>CTI Services</td>
<td>Cisco IP Manager Assistant</td>
</tr>
<tr>
<td></td>
<td>Cisco WebDialer Web Service</td>
</tr>
<tr>
<td>CDR Services</td>
<td>Cisco SOAP - CDRonDemand Service</td>
</tr>
<tr>
<td></td>
<td>Cisco CAR Web Service</td>
</tr>
<tr>
<td>Database and Admin Services</td>
<td>Cisco Bulk Provisioning Service</td>
</tr>
<tr>
<td></td>
<td>AXL Web Service</td>
</tr>
<tr>
<td></td>
<td>Cisco URL Web Service</td>
</tr>
<tr>
<td>Performance and Monitoring Services</td>
<td>Cisco Serviceability Reporter</td>
</tr>
<tr>
<td></td>
<td>Cisco Certificate Authority Proxy Function</td>
</tr>
<tr>
<td>Directory Services</td>
<td>Cisco DirSync</td>
</tr>
</tbody>
</table>

### Tip
You can safely disable the following services if you do not plan to use them:

- Cisco Messaging Interface
- Cisco DHCP Monitor Service
- Cisco TAPS Service
- Cisco Directory Number Alias Sync
- Cisco Directory Number Alias Sync
- Cisco Dialed Number Analyzer Server
- Cisco Dialed Number Analyzer
- Self Provisioning IVR

---

**Configure Initial Parameters for the System**

**Recommended Services for Publisher Nodes**
Recommended Services for Subscriber Nodes

The following table lists recommended services for a Unified Communications Manager subscriber node when using a non-dedicated TFTP server.

Tip

You can safely disable the other services if you don't plan to use them.

Table 6: Recommended Subscriber Node Services for Non-Dedicated TFTP Server Deployments

<table>
<thead>
<tr>
<th>Type</th>
<th>Service Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>CM Services</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td></td>
<td>Cisco IP Voice Media Streaming App</td>
</tr>
<tr>
<td></td>
<td>Cisco CTIManager</td>
</tr>
<tr>
<td></td>
<td>Cisco Extension Mobility</td>
</tr>
<tr>
<td></td>
<td>Cisco Extended Functions</td>
</tr>
<tr>
<td></td>
<td>Cisco TFTP</td>
</tr>
</tbody>
</table>

You must activate the following services on each Instant Messaging and Presence node in your cluster.

- Cisco SIP Proxy
- Cisco Presence Engine
- Cisco XCP Connection Manager
- Cisco XCP Authentication Service

Configure Service Parameters

You can configure the service parameters on the node using Cisco Unified Communications Manager Administration. Service parameters that are marked as cluster-wide affect all nodes in the cluster.

Caution

Some changes to service parameters can cause system failure. We recommend that you do not make any changes to service parameters unless you fully understand the feature that you are changing or unless the Cisco Technical Assistance Center (TAC) specifies the changes.

Before you begin

- Make sure that the Unified Communications Manager nodes are configured.
- Make sure that the service is active. For details, see Activate Essential Services, on page 36.
Configure Initial Parameters for the System

View Clusterwide Service Parameter Settings

Procedure

**Step 1** From Cisco Unified CM Administration, choose choose System > Service Parameters.

**Step 2** Select a node in the Server drop-down list.

**Step 3** Select a service in the Service drop-down list.

**Tip** Click the ? icon in the Service Parameter Configuration window to view a list of service parameters along with their descriptions.

**Step 4** Click Advanced to view the full list of parameters.

**Step 5** Modify the service parameters and then click Save.

The window refreshes and the service parameter values are updated.

You can click the Set to Default button to update all parameters to the suggested value that appears after the Parameter Value field. If a parameter does not have a suggested value, the service parameter value does not change when you click the Set to Default button.

---

View Clusterwide Service Parameter Settings

You can use Cisco Unified Communications Manager Assistant and Cisco Unified Serviceability to view the status of services for nodes in your cluster. To view service parameter settings and parameter descriptions, use Cisco Unified Communications Manager Assistant.

Procedure

**Step 1** To display services and view service parameter settings for a node using Cisco Unified Communications Manager Assistant, perform the following steps.

a) Select System > Service Parameters.

b) In the Service Parameters Configuration window, select a node in the Server drop-down box.

c) Select a service in the Service drop-down box.

All parameters that apply to the selected node appear. Parameters that appear in the Clusterwide Parameters (General) section apply to all nodes in the cluster.

d) Click the (?) icon in the Service Parameter Configuration window to view a list of service parameters along with their descriptions.

**Step 2** To display the service parameters for a particular service on all nodes in a cluster, select Parameters for All Servers in the Related Links drop-down box in the Service Parameters Configuration window, then click Go.

The Parameters for All Servers window appears. You can click on a server name that is listed or on a parameter value to open the related Service Parameter Configuration window.

**Step 3** To display out-of-sync service parameters for a particular service on all nodes in a cluster, select Out of Sync Parameters for All Servers in the Related Links drop-down box in the Parameters for All Servers window, then click Go.
The Out of Sync Parameters for All Servers window appears. You can click on a server name that is listed or on a parameter value to open the related Service Parameter Configuration window.
Device Pools Overview

Device pools provide a common set of configurations for a group of devices. You can assign a device pool to devices such as phones, gateways, trunks and CTI route points. After you create a device pool, you can associate devices so that they inherit the device pool settings, rather than configuring each device individually.

Device pools let you configure devices according to their location, by assigning location-related information such as Date/Time Groups, Regions, and Phone NTP References. You can create as many device pools as you need, typically one per location. However, you can also apply device pools to apply configurations according to a job function (for example, if your company has a call center, you may want to assign call center phones to one device pool and administration office phones to another).

This section covers the steps that are required to set up core settings for device pools, such as:

- Network Time Protocol—Configure Phone NTP References to provide NTP support for SIP devices in the device pool.
- Regions—Manage bandwidth and supported audio codecs for calls to and from certain regions.
- Cisco Unified Communications Manager Groups—Configure call processing redundancy and distributed call processing for your devices.

Network Time Protocol Overview

The Network Time Protocol (NTP) allows network devices such as SIP phones to synchronize their clocks to a network time server or network-capable clock. NTP is critical for ensuring that all network devices have the same time and that the timestamps in audit logs match the network time. Features such as billing and call detail records rely on accurate timestamps across the network. In addition, system administrators need accurate timestamps in audit logs for troubleshooting. This allows them to compare audit logs from different systems and create a reliable timeline and sequence of events for whatever issue they are facing.
During installation, you must set up an NTP server for the Unified Communications Manager publisher node. The subscriber nodes then sync their time from the publisher node.

You can assign up to five NTP servers.

**Phone NTP References**

- **For SIP Phones**: It is mandatory that you configure Phone NTP References and assign them through the device pool. These references direct the SIP phone to an appropriate NTP server that can provide the network time. If a SIP phone cannot get its date/time from the provisioned “Phone NTP Reference” the phone receives this information when it registers with Unified Communications Manager.

- **For SCCP Phones**: Phone NTP References are not required as SCCP phones obtain their network time from Unified Communications Manager directly through SCCP signaling.

**Regions Overview**

Regions provide capacity controls for Unified Communications Manager multi-site deployments where you may need to limit the bandwidth for certain calls. For example, you can use regions to limit the bandwidth for calls that are sent across a WAN link, while maintaining a higher bandwidth for internal calls. You can use regions to limit the bandwidth for audio and video calls by setting the maximum bitrate for intraregional or interregional calls to whatever the region(s) can provide.

Additionally, the system uses regions to set the audio codec priority where you have applications that support specific codecs only. You can configure a prioritized list of supported audio codecs and apply it to calls to and from specific regions.

When you configure the maximum audio bit rate setting in the Region Configuration window (or use the service parameter in the Service Parameter Configuration window), this setting serves as a filter. When an audio codec is selected for a call, Unified Communications Manager takes the matching codecs from both sides of a call leg, filters out the codecs that exceed the configured maximum audio bit rate, and then picks the preferred codec among the codecs that are remaining in the list.

Unified Communications Manager supports up to 2000 regions.

**Supported Audio Codecs**

Unified Communications Manager supports video stream encryption and the following audio codecs:

<table>
<thead>
<tr>
<th>Audio Codec</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>The most commonly supported codec, used over the public switched telephone network.</td>
</tr>
<tr>
<td>G.722</td>
<td>Wideband codec often used in video conferences. This is always preferred by Unified Communications Manager over G.711, unless G.722 is disabled.</td>
</tr>
<tr>
<td>G.722.1</td>
<td>Low complexity wideband codec operating at 24 and 32 kb/s. The audio quality approaches that of G.722 while using, at most, half the bit rate.</td>
</tr>
<tr>
<td>G.728</td>
<td>Low bit rate codec that video endpoints support.</td>
</tr>
<tr>
<td>G.729</td>
<td>Low bit rate codec with 8 kb/s compression that is supported by Cisco IP Phone 7900, and typically used for calls across a WAN link.</td>
</tr>
<tr>
<td>Audio Codec</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>GSM</td>
<td>The global system for mobile communications (GSM) codec. GSM enables the MNET system for GSM wireless handsets to operate with Unified Communications Manager.</td>
</tr>
<tr>
<td>L16</td>
<td>Advanced Audio Coding-Low Delay (AAC-LD) is a super-wideband audio codec that provides superior sound quality for voice and music. This codec provides equal or improved sound quality over older codecs, even at lower bit rates.</td>
</tr>
<tr>
<td>AAC-LD (mpeg4-generic)</td>
<td>Supported for SIP devices, in particular, Cisco TelePresence systems.</td>
</tr>
<tr>
<td>AAC-LD (MP4A-LATM)</td>
<td>Low-overhead MPEG-4 Audio Transport Multiplex (LATM) is a super-wideband audio codec that provides superior sound. Supported for SIP devices including Tandberg and some third-party endpoints.</td>
</tr>
<tr>
<td>Note</td>
<td>AAC-LD (mpeg4-generic) and AAC-LD (MP4A-LATM) are not compatible.</td>
</tr>
<tr>
<td>Internet Speech Audio Codec (iSAC)</td>
<td>An adaptive wideband audio codec, specially designed to deliver wideband sound quality with low delay in both low and medium bit rate applications.</td>
</tr>
<tr>
<td>Internet Low Bit Rate Codec (iLBC)</td>
<td>Provides audio quality between G.711 and G.729 at bit rates of 15.2 and 13.3 kb/s while allowing for graceful speech quality degradation in a lossy network due to independently encoded speech frames. iLBC is supported for SIP, SCCP, H323, and MGCP devices.</td>
</tr>
<tr>
<td>Note</td>
<td>H.323 Outbound FastStart does not support the iLBC codec.</td>
</tr>
<tr>
<td>Adaptive Multi-Rate (AMR)</td>
<td>The required standard codec for 2.5G/3G wireless networks based on GSM (WDMA, EDGE, GPRS). This codec encodes narrowband (200-3400 Hz) signals at variable bit rates ranging from 4.75 to 12.2 kb/s with toll quality speech starting at 7.4 kb/s. AMR is supported only for SIP devices.</td>
</tr>
<tr>
<td>Adaptive Multi-Rate Wideband (AMR-WB)</td>
<td>Codified as G.722.2, an ITU-T standard speech codec formally known as Wideband, codes speech at about 16 kb/s. This codec is preferred over other narrowband speech codecs such as AMR and G.711 because it provides better speech quality due to a wider speech bandwidth of 50 Hz to 7000 Hz. AMR-WB is supported only for SIP devices.</td>
</tr>
</tbody>
</table>
Opus codec is an interactive speech and audio codec, specially designed to handle a wide range of interactive audio applications such as voice over IP, video conferencing, in-game chat, and live distributed music performance. This codec scales from narrowband low bit rate to a very high quality bit rate ranging from 6 to 510 kb/s.

Opus codec support is enabled by default for all SIP devices. You can reconfigure Opus support via the Opus Codec Enabled service parameter (the default setting is Enabled for All Devices). You can reconfigure this parameter to disable Opus codec support, or to enable support in non-recording devices only.

**Note**

Opus has a dependency on the G.722 codec. The Advertise G.722 Codec enterprise parameter should also be set to Enabled for SIP devices to use Opus.

---

**Cisco Unified CM Groups Overview**

A Unified Communications Manager Group is a prioritized list of up to three redundant servers to which devices can register. Each group contains a primary node and up to two backup nodes. The order in which you list the nodes determines their priority with the first node being the primary node, the second being the backup node, and the third being the tertiary node. You can assign a device to a Cisco Unified Communications Manager Group via the **Device Pool Configuration**.

Unified Communications Manager groups provide two important features for your system:

- **Call processing redundancy**—When a device registers, it attempts to connect to the primary (first) Unified Communications Manager in the group that is assigned to its device pool. If the primary Unified Communications Manager is not available, the device tries to connect to the first backup node and if that node is unavailable, it tries to connect to the tertiary node. Each device pool has one Unified Communications Manager group that is assigned to it.

- **Distributed call processing**—You can create multiple device pools and Unified Communications Manager groups to distribute device registrations evenly across multiple Unified Communications Managers.

For most systems, you will assign a single Unified Communications Manager to multiple groups to achieve better load distribution and redundancy.

**Call Processing Redundancy**

Unified Communications Manager groups provide call processing redundancy and recovery:

- **Failover**—Occurs when the primary Unified Communications Manager in a group fails, and the devices reregister with the backup Unified Communications Manager in that group.

- **Fallback**—Occurs when a failed primary Unified Communications Manager comes back into service, and the devices in that group reregister with the primary Unified Communications Manager.

Under normal operation, the primary Unified Communications Manager in a group controls call processing for all the registered devices (such as phones and gateways) that are associated with that group.
If the primary Unified Communications Manager fails for any reason, the first backup Unified Communications Manager in the group takes control of the devices that were registered with the primary Unified Communications Manager. If you specify a second backup Unified Communications Manager for the group, it takes control of the devices if both the primary and the first backup Unified Communications Managers fail.

When a failed primary Unified Communications Manager comes back into service, it takes control of the group again, and the devices in that group automatically reregister with the primary Unified Communications Manager.

Example

For example, the following figure shows a simple system with three Unified Communications Managers in a single group that is controlling 800 devices.

**Figure 1: Unified Communications Manager Group**

The figure depicts Unified Communications Manager group G1 that is assigned with two device pools, DP1 and DP2. Unified Communications Manager 1, as the primary Unified Communications Manager in group G1, controls all 800 devices in DP1 and DP2 under normal operation. If Unified Communications Manager 1 fails, control of all 800 devices transfers to Unified Communications Manager 2. If Unified Communications Manager 2 also fails, control of all 800 devices transfers to Unified Communications Manager 3.

The configuration provides call-processing redundancy, but it does not distribute the call-processing load very well among the three Unified Communications Managers in the example. Refer to the following topic for information on how to use Unified Communications Manager groups and device pools to provide distributed call processing within the cluster.

---

**Note**

Empty Unified Communications Manager groups will not function.

### Distributed Call Processing

Unified Communications Manager groups provide both call-processing redundancy and distributed call processing. How you distribute devices, device pools, and Unified Communications Managers among the groups determines the level of redundancy and load balancing in your system.

In most cases, you would want to distribute the devices in a way that prevents the other Unified Communications Managers from becoming overloaded if one Unified Communications Manager in the group fails. The following figure shows one possible way to configure the Unified Communications Manager groups and device pools.
to achieve both distributed call processing and redundancy for a system of three Unified Communications Managers and 800 devices.

Figure 2: Redundancy Combined with Distributed Call Processing

The previous figure depicts the Unified Communications Manager groups as they are configured and assigned to device pools, so Unified Communications Manager 1 serves as the primary controller in two groups, G1 and G2. If Unified Communications Manager 1 fails, the 100 devices in device pool DP1 reregister with Unified Communications Manager 2, and the 300 devices in DP2 reregister with Unified Communications Manager 3. Similarly, Unified Communications Manager 2 serves as the primary controller of groups G3 and G4. If Unified Communications Manager 2 fails, the 100 devices in DP3 reregister with Unified Communications Manager 1, and the 300 devices in DP4 reregister with Unified Communications Manager 3. If Unified Communications Manager 1 and Unified Communications Manager 2 both fail, all devices reregister with Unified Communications Manager 3.
Device Pool Prerequisites

Make sure to properly plan out your device pools before you configure them. When configuring device pools and redundant Unified Communications Manager Groups, you will want to provide server redundancy for phones while distributing registrations evenly across your cluster. For additional information that you can use to plan your system, refer to the Cisco Collaboration System Solution Reference Network Design at https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-implementation-design-guides-list.html.

To ensure that Unified Communications Manager includes the latest time zone information, you can install a Cisco Options Package (COP) file that updates the time zone information after you install Unified Communications Manager. After major time zone change events, we will contact you to let you know that you can download the latest COP file at https://software.cisco.com/download/navigator.html.

Change the settings for CMLocal to your local date and time.

Additional Device Pool Configurations

This chapter focuses on core settings such as phone NTP references, regions and call processing redundancy via Unified Communications Manager Groups. However, you can also apply these optional features and components to devices via the device pool configuration:

- Media Resources—Assign media resources such as conference bridges, and music on hold to the devices in your device pool. Refer to the "Configure Media Resources" section of this book for details on configuring media resources. See Media Resources Configuration Task Flow, on page 435.

- Survivable Remote Site Telephony (SRST)—If your deployment uses WAN connections, configure SRST so that in the event of a WAN outage, IP gateways can provide limited call support. See Survivable Remote Site Telephony Configuration Task Flow, on page 104.

- Call Routing Information—For information on how to route calls between clusters, see Call Routing Configuration Task Flow, on page 126.

- Device Mobility—Configure Device Mobility groups to allow devices to assume the settings based on their physical location. For details, see the "Configure Device Mobility" chapter in the Feature Configuration Guide for Cisco Unified Communications Manager.

Core Settings for Device Pools Configuration Task Flow

Complete these tasks to set up device pools and apply settings such as Regions, Phone NTP references, and redundancy for the devices that use those device pools.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Configure the Network Time Protocol, on page 48</td>
<td>Complete the tasks in this task flow to set up NTP on your system. Configure Phone NTP references and apply them to a Date/Time Group that you can assign to a device pool.</td>
</tr>
</tbody>
</table>
## Configure the Network Time Protocol

Complete these tasks to configure the Network Time Protocol (NTP) for your system. Configure Phone NTP References and apply them to a Date/Time Group which you can then apply to a device pool.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
</tbody>
</table>
| Add an NTP Server, on page 49 | Optional. Use this procedure if you need to add an NTP server. You can add up to five NTP servers.  
**Note** During system installation, you were required to point Unified Communications Manager to an NTP server. You can use this procedure if you want to add additional NTP servers. Otherwise, you can skip this task. |
| **Step 2** | |  |
| Configure Phone NTP References, on page 49 | For SIP phones, it's mandatory that you configure phone NTP references and then apply them via a Date/Time Group and Device Pool. |
| **Step 3** | |  |
| Add a Date/Time Group, on page 50 | Define time zones for the various devices that are connected to your system and assign the Phone NTP references that you've set up to the appropriate Date/Time Group. |
For additional information on CLI commands that you can use to troubleshoot and configure NTP such as the `utils ntp` set of commands, refer to the *Command Line Interface Reference Guide* at https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

**Add an NTP Server**

Add an NTP Server to Unified Communications Manager.

**Procedure**

- **Step 1** Log in to the Command Line Interface.
- **Step 2** To confirm that the publisher node can reach the NTP server, run the `utils network ping <ip_address>` where the `ip_address` represents the address of the NTP server.
- **Step 3** If the server is reachable, run the `utils ntp server add <ip_address>` to add the server.
- **Step 4** Restart the NTP service with the `utils ntp restart` command.

**Configure Phone NTP References**

Use this procedure to configure Phone NTP References, which are mandatory for SIP phones. You can assign the NTP references that you create to a device pool via the Date/Time Group. The reference points the SIP phone to an appropriate NTP server that can provide the network time. For SCCP phones, this configuration is not required.

**Note**

Unified Communications Manager does not support the multicast and anycast modes. If you choose either of these modes, your system defaults to the directed broadcast mode.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose System > Phone NTP Reference.
- **Step 2** Click Add New.
- **Step 3** In the IP Address field, enter the IP Address of the NTP server that you want the phone that is running SIP to use to get its date and time.
- **Step 4** In the Description field, enter a description for the phone NTP reference.
Step 5 From the Mode drop-down list, choose the mode for the phone NTP reference from the following list of options:

- **Unicast**—If you choose this mode, the phone sends an NTP query packet to that particular NTP server.
- **Directed Broadcast**—If you choose this default NTP mode, the phone accesses date/time information from any NTP server but gives the listed NTP servers (1st = primary, 2nd = secondary) priority.

**Note** Cisco TelePresence device types support Unicast mode only.

Step 6 Click Save.

---

**What to do next**

Assign the Phone NTP Reference(s) to a Date/Time Group. For details, see Add a Date/Time Group, on page 50

---

**Add a Date/Time Group**

Configure Date/Time Groups to define time zones in your system. Assign the Phone NTP references that you configured to the appropriate group. After adding a new date/time group to the database, you can assign it to a device pool to configure the date and time information for all devices in that device pool.

You must reset devices to apply any changes that you make.

**Tip** For a worldwide distribution of Cisco IP Phones, create a date/time group for each time zones.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose System > Date/Time Group.

**Step 2** Click Add New.

**Step 3** Assign NTP References to this group:

a) Click Add Phone NTP References.

b) In the Find and List Phone NTP References popup, click Find and select the phone NTP reference(s) that you configured in the previous task.

c) Click Add Selected.

d) If you added multiple references, use the up and down arrows to changed the prioritized order. The references at the top have the higher priority.

**Step 4** Configure the remaining fields in the Date/Time Group Configuration window. See the online help for more information about the fields and their configuration options.

**Step 5** Click Save.
Configure Regions

Complete the following tasks to configure regions for your device pools. Configure relationships between regions to better manage bandwidth. You can use Regions to control the maximum bit rates for certain types of calls (for example, video calls) and to prioritize specific audio codecs.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Customize Audio Codec Preferences, on page 51</td>
<td>Optional. Use this procedure if you want to customize priorities for your audio codecs. You may want to do this in order to prioritize specific audio codecs ahead of other codecs. Otherwise, you can assign one of the default audio codec lists to your device pools.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Clusterwide Defaults for Regions, on page 52</td>
<td>Configure the clusterwide defaults for Regions. All Regions will use these default settings unless you configure otherwise within the Region Configuration.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Region Relationships, on page 52</td>
<td>Set up new regions or edit settings for existing regions. Configure relationships for both interregional and intraregional calls.</td>
</tr>
</tbody>
</table>

Customize Audio Codec Preferences

Use this procedure to customize priorities for your audio codecs. Create a new audio codec preferences list by copying settings from an existing list, and then editing the order of priority within your new list.

Note

If you don't need to customize audio codec priorities, you can skip this task. When you configure your device pools, you can assign one of the default audio codec preference lists.

Procedure

Step 1 From Cisco Unified CM Administration choose System > Region Information > Audio Codec Preference List.
Step 2 Click Add New.
Step 3 From the Audio Codec Preference Lists drop-down list box, select one of the existing audio codec preference lists. The prioritized list of audio codecs displays for the list that you selected.
Step 4 Click Copy. The prioritized list of codecs from the copied list is applied to a newly created list.
Step 5 Edit the Name for your new audio codec list. For example, customizedCodecList.
Step 6 Edit the Description.
Configure Clusterwide Defaults for Regions

Use this procedure to configure default settings clusterwide for Regions. These settings apply by default to calls to and from all regions unless you configure region relationships for individual regions within the Region Configuration window.

Procedure

1. From Cisco Unified CM Administration choose System > Service Parameters.
2. From the Server drop-down list, select a Unified Communications Manager node.
3. From the Service drop-down list, select the Cisco CallManager service.
   The Service Parameter Configuration window displays.
4. Under Clusterwide Parameters (System - Location and Region), configure any new service parameter settings that you want. For service parameter descriptions, click any of the parameter names to view the help description.
5. Click Save.

Configure Region Relationships

Use this procedure to create Regions and to assign custom settings for calls between specific regions. You can edit settings such as preferred audio codecs and maximum bitrates. For example, if you have a region with lower bandwidth capacities than the rest of the network, you may want to edit the maximum session bit rate for video calls to and from the region. You could reset this value to whatever that region can provide.

Note
For enhanced scalability, and to ensure that the system uses fewer resources, we recommend that you use the default values from the Service Parameters Configuration window wherever possible.

Procedure

1. From Cisco Unified CM Administration, choose System > Region Information > Regions.
2. Do either of the following:
   • Click Find and select a region.
   • Click Add New to create a new region.
   • Enter a Name for the Region. For example, New York.
• Click Save.

The read-only Region Relationships area displays any customized settings that you've set up between the selected region and another region.

**Step 3**

To modify the settings between this region and another region (or the same region for intraregional calls), edit the settings in the **Modify Relationships to other Regions** area:

a) In the Regions area, highlight the other region (for intraregional calls, highlight the same region that you are configuring).

b) Edit the settings in the adjacent fields. For help with the fields and their settings, see the online help.

c) Click Save.

The new settings now display as a custom rule in the Region Relationships area.

**Note**

If you edit a region relationship within one region there is no need to duplicate that configuration in the other region as the settings will update in the other region automatically. For example, let's say that you open Region 1 in the Region Configuration window and configure a custom relationship to Region 2. If you were to then open Region 2, you would see the custom relationship displayed in the Region Relationships area.

---

## Configure Cisco Unified CM Groups

Use this procedure to set up Unified Communications Manager Groups for call processing redundancy, load balancing and failover for the devices in the device pool.

**Tip**

Set up multiple groups and device pools where the primary server is different in each group so as to provide distributed call processing where device registrations are balanced evenly across the cluster nodes.

**Note**

Do not use the default server group because it is not descriptive and can cause confusion.

**Procedure**

**Step 1**

From Cisco Unified CM Administration, choose **System > Cisco Unified CM Group**.

**Step 2**

Enter a **Name** for the group.

**Note**

Consider identifying the order of the nodes in the name so that you can easily distinguish the group from others. For example, CUCM_PUB-SUB.

**Step 3**

Check the **Auto-registration Cisco Unified Communications Manager Group** check box if you want this Unified Communications Manager group to be the default Unified Communications Manager group when auto-registration is enabled.

**Step 4**

From the **Available Cisco Unified Communications Managers** list, choose the nodes that you want to add to this group, and click the down arrow to select them. You can add up to three servers to a group.
Configure Device Pools

The servers in this group appear in the **Selected Cisco Unified Communications Managers** list box. The top server in the list is the primary server.

**Step 5**
Use the arrows beside the **Selected Cisco Unified Communications Managers** list box to change which servers are the primary, and backup servers.

**Step 6**
Click **Save**.

---

**Configure Device Pools**

Set up device pools for your system devices. Apply the other core settings that you configured to the device pools in order to apply those settings to the devices that use this device pool. You can configure multiple device pools to meet your deployment needs.

**Before you begin**
If you want to assign an SRST configuration, refer to *Survivable Remote Site Telephony Configuration Task Flow*, on page 104.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **System > Device Pool**.

**Step 2**
Do either of the following:
- Click **Add New** to create a new device pool.
- Click **Find** and select an existing device pool.

**Step 3**
In the **Device Pool Name** field, enter a name for the device pool.

**Step 4**
From the **Cisco Unified Communications Manager Group** drop-down, select the group that you set up to handle call processing redundancy and load balancing.

**Step 5**
From the **Date/Time Group** drop-down, select the group that you set up to handle date, time, and phone NTP references for the devices that use this device pool.

**Step 6**
From the **Region** drop-down list box, select the region that you want to apply to this device pool.

**Step 7**
From the **Media Resource Group List** drop-down, select a list that contains the media resources that you want to apply to this device pool.

**Step 8**
Apply SRST settings for this device pool:
- **a)** From the **SRST Reference** drop-down, assign an SRST reference.
- **b)** Assign a value for the **Connection Monitor Duration** field. This setting defines the time that the phone monitors its connection to Unified Communications Manager before it unregisterring from SRST and reregisterring to Unified Communications Manager.

**Step 9**
Complete the remaining fields in the **Device Pool Configuration** window. For help with the fields and their settings, see the online help.

**Step 10**
Click **Save**.
Configure multiple device pools according to your deployment requirements.

**Basic Device Pool Configuration Fields**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Pool Name</td>
<td>Enter the name of the new device pool. You can enter up to 50 characters, which include alphanumeric characters, periods (.), hyphens (-), underscores (_), and blank spaces.</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group</td>
<td>Choose the Cisco Unified Communications Manager group to assign to devices in this device pool. A Cisco Unified Communications Manager group specifies a prioritized list of up to three Unified Communications Manager nodes. The first node in the list serves as the primary node for that group, and the other members of the group serve as backup nodes for redundancy.</td>
</tr>
<tr>
<td>Date/Time Group</td>
<td>Choose the date/time group to assign to devices in this device pool. The date/time group specifies the time zone and the display formats for date and time.</td>
</tr>
<tr>
<td>Region</td>
<td>Choose the region to assign to devices in this device pool. The region settings specify voice and video codecs that can be used for communications within a region and between other regions.</td>
</tr>
</tbody>
</table>

**Call Preservation**

The call preservation feature of Unified Communications Manager ensures that an active call does not get interrupted when a Unified Communications Manager fails or when communication fails between the device and the Unified Communications Manager that set up the call.

Unified Communications Manager supports full call preservation for an extended set of Cisco Unified Communications devices. This support includes call preservation between Cisco Unified IP Phones, Media Gateway Control Protocol (MGCP) gateways that support Foreign Exchange Office (FXO) (non-loop-start trunks) and Foreign Exchange Station (FXS) interfaces, and, to a lesser extent, conference bridge, MTP, and transcoding resource devices.

Enable H.323 call preservation by setting the advanced service parameter, Allow Peer to Preserve H.323 Calls, to True.

The following devices and applications support call preservation. If both parties connect through one of the following devices, Unified Communications Manager maintains call preservation:

- Cisco Unified IP Phones
- SIP trunks
- Software conference bridge
- Software MTP
• Hardware conference bridge (Cisco Catalyst 6000 8 Port Voice E1/T1 and Services Module, Cisco Catalyst 4000 Access Gateway Module)

• Transcoder (Cisco Catalyst 6000 8 Port Voice E1/T1 and Services Module, Cisco Catalyst 4000 Access Gateway Module)

• Non-IOS MGCP gateways (Catalyst 6000 24 Port FXS Analog Interface Module, Cisco DT24+, Cisco DE30+, Cisco VG200)

• Cisco IOS H.323 gateways (such as Cisco 2800 series, Cisco 3800 series)

• Cisco IOS MGCP Gateways (Cisco VG200, Catalyst 4000 Access Gateway Module, Cisco 2620, Cisco 3620, Cisco 3640, Cisco 3660, Cisco 3810)

• Cisco VG248 Analog Phone Gateway

The following devices and applications do not support call preservation:

• Annunciator

• H.323 endpoints such as NetMeeting or third-party H.323 endpoints

• CTI applications

• TAPI applications

• JTAPI applications

## Call Preservation Scenarios

Below table describes how call preservation is handled in various scenarios.

### Table 8: Call Preservation Scenarios

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Call Preservation Handling</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unified Communications Manager fails.</td>
<td>A Unified Communications Manager failure causes the call-processing function for all calls that were set up through the failed Unified Communications Manager to be lost. Unified Communications Manager maintains affected active calls until the end user hangs up or until the devices can determine that the media connection has been released. Users cannot invoke any call-processing features for calls that are maintained as a result of this failure.</td>
</tr>
<tr>
<td>Scenario</td>
<td>Call Preservation Handling</td>
</tr>
<tr>
<td>----------</td>
<td>---------------------------</td>
</tr>
</tbody>
</table>
| Communication failure occurs between Unified Communications Manager and device. | When communication fails between a device and the Unified Communications Manager that controls it, the device recognizes the failure and maintains active connections. The Unified Communications Manager recognizes the communication failure and clears call-processing entities that are associated with calls in the device where communication was lost.

The Unified Communications Managers still maintain control of the surviving devices that are associated with the affected calls. Unified Communications Manager maintains affected active calls until the end user hangs up or until the devices can determine that the media connection has been released. Users cannot invoke any call-processing features for calls that are maintained as a result of this failure.

**Note** In case of a failover, when you bring up the Unified CM node within the KeepAlive timer, the phone remains registered to the current node even though the call is in preservation mode. This is possible as KeepAlive time is active. |
| Device failure (Phone, gateway, conference bridge, transcoder, MTP) | When a device fails, the connections that exist through the device stop streaming media. The active Unified Communications Manager recognizes the device failure and clears call-processing entities that are associated with calls in the failed device.

The Unified Communications Manager maintain control of the surviving devices that are associated with the affected calls. Unified Communications Manager maintains the active connections (calls) that are associated with the surviving devices until the surviving end users hang up or until the surviving devices can determine that the media connection has been released. |
PART II

Enable Inbound and Outbound Calling

• Inbound Outbound Calling Overview, on page 61
• Configure Gateways, on page 63
• Configure SIP Normalization and Transparency, on page 79
• Configure SDP Transparency Profiles, on page 85
• Configure SIP Profiles, on page 87
• Configure Dual Stack IPv6, on page 89
• Configure SIP Trunks, on page 95
• Configure H.323 Trunks, on page 101
• Configure SRST, on page 103
## Inbound Outbound Calling Overview

- About Inbound and Outbound Calling, on page 61
- Inbound and Outbound Calling Configuration, on page 61

### About Inbound and Outbound Calling

This part describes how to set up the inbound and outbound calling for your system.

### Inbound and Outbound Calling Configuration

Complete the following task flows to configure Inbound and Outbound Calling for your system.

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Gateway Configuration Task Flow, on page 66</td>
<td>Add gateways to your system.</td>
</tr>
<tr>
<td>Step 2</td>
<td>SIP Normalization and Transparency Configuration Task Flow, on page 80</td>
<td>Optional. Configure SIP Normalization and transparency scripts that you can assign to SIP trunks or SIP devices in order to resolve SIP interoperability issues.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure SDP Transparency Profile, on page 86</td>
<td>Optional. If your SIP deployment requires support for SDP attributes that are not natively supported by Unified Communications Manager, set up an SDP transparency profile that includes the non-supported attributes.</td>
</tr>
<tr>
<td>Step 4</td>
<td>SIP Profile Overview, on page 87</td>
<td>Configure SIP profiles for your SIP trunks and SIP devices.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Dual Stack IPv6 Configuration Task Flow, on page 90</td>
<td>Optional. If your SIP deployment requires support for IPv6 devices, configure dual stack IPv6 support in your system. Dual stack can be configured for SIP deployments only.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------------------------------------------</td>
<td>----------------------------------------------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure SIP trunks for your system.</td>
<td></td>
</tr>
<tr>
<td>SIP Trunk Configuration Task Flow, on page 97</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Configure H.323 trunks for your system.</td>
<td></td>
</tr>
<tr>
<td>H.323 Trunk Overview, on page 101</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Configure your system for SRST.</td>
<td></td>
</tr>
<tr>
<td>Survivable Remote Site Telephony Configuration Task Flow, on page 104</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
CHAPTER 9

Configure Gateways

• Gateway Overview, on page 63
• Gateway Setup Prerequisites, on page 65
• Gateway Configuration Task Flow, on page 66

Gateway Overview

Cisco offers a wide variety of voice and video gateways. A gateway provides interfaces that allow the Unified Communications network to communicate with an external network. Traditionally, gateways have been used to connect the IP-based Unified Communications network to legacy telephone interfaces such as the PSTN, a private branch exchange (PBX), or legacy devices such as an analog phone or fax machine. In its simplest form, a voice gateway has an IP interface and a legacy telephony interface, and the gateway translates messages between the two networks so that the two networks can communicate.

Gateway Protocols

Most Cisco gateways offer multiple deployment options and can be deployed using any one of a number of protocols. Depending on the gateway that you want to deploy, your gateway may be configurable using any of the following communication protocols:

• Media Gateway Control Protocol (MGCP)
• Skinny Call Control Policy (SCCP)
• Session Initiation Protocol (SIP)
• H.323

Vendor Interface Cards

The Vendor Interface Card (VIC) must be installed on the gateway to provide a connection interface for external networks. Most gateways offer multiple VIC options and each VIC may offer many different ports and connection types for both analog and digital connections.

Refer to your gateway documentation for the protocols, cards, and connections that are offered with your gateway.
Port and Trunk Connection Types

Following are the main types of port connections that you can configure on gateways:

- Foreign Exchange Station (FXS)—FXS ports offer connections to analog stations such as an analog phone, speakerphone, or legacy voicemail system.
- Foreign Exchange Office (FXO)—FXO ports offer analog connections to the PSTN or a legacy PBX.
- T1 Channel Associated Signaling (T1/E1 CAS)—T1/E1 CAS connections offer digital trunk connections to a central office, PBX, or other analog device.
- Primary Rate Interface (T1/E1 PRI)—Digital access PRI connections are widely used in corporate communications. T1 PRI is widely used in North America and Japan and offers 23 B-channels for voice and data and one D-channel for common channel signaling at a rate of 1.544 Mb/s. E1 is widely used in Europe, offering 30 B-channels for voice and data, one D-channel for common signaling, and one framing channel. E1 uses a rate of 2.048 Mb/s.
- Basic Rate Interface (BRI)—BRI is a digital telephony protocol, which is used for small office and home communications links, provides two B-channels for voice and data and one D-channel for signaling.

Connection Types per Protocol

MGCP gateways offer the following connection types:

- TI/E1 PRI Digital access
- T1 CAS
- BRI
- FXO
- FXS

SCCP gateways offer the following connection types:

- FXS
- BRI

SIP gateways offer the following connections:

- FXS
- FXS-DID
- E&M
- BRI
- BRI QSIG
- T1 CAS
- T1 FGD
- E1 CAS
Enable Inbound and Outbound Calling

Gateway Setup Prerequisites

Gateway Setup Prerequisites

Install the Hardware

Before you configure the gateway in Cisco Unified Communications Manager, you must perform the following tasks on your gateway hardware:

- Install and configure the gateway
- Install any vendor interface cards (VICs) on the gateway.
- Use the CLI to configure IOS on the gateway.

For details, refer to the hardware and software documentation that comes with your gateway.
To get to the default web pages for many gateway devices, you can use the IP address of that gateway. Make your hyperlink url = http://x.x.x.x/, where x.x.x.x is the dot-form IP address of the device. The web page for each gateway contains device information and the real-time status of the gateway.

**Plan the Gateway Deployment**

Before configuring the gateway in Cisco Unified Communications Manager, make sure that you adequately plan the types of connections that you want to configure on the gateway. Many gateways can be configured using any one of MGCP, SIP, H.323, or SCCP as the gateway protocol. The connection types for each type of deployment vary according to the protocol that you choose and the VICs that are installed on the gateway. Be sure to understand the following:

- Which gateway protocols does your gateway support.
- What types of port connections the VICs on the gateway support.
- What types of connections are you planning on configuring?
- For analog connections, are you connecting to the PSTN, legacy PBX, or to legacy devices.
- For digital access connections, are you connecting to a T1 CAS interface, or to a PRI interface?
- For FXO connections, how do you want to direct incoming calls? Are you directing incoming calls to an automated IVR or to an attendant?

**Gateway Configuration Task Flow**

Perform the following tasks to add your network gateways to Unified Communications Manager.

**Before you begin**

Review the Gateway Setup Prerequisites, on page 65.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configure your gateways in Unified Communications Manager. Perform any of the following procedures depending on the protocol that you want to deploy:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Configure MGCP Gateway, on page 67</td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Configure SCCP Gateway, on page 74</td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Configure SIP Gateway, on page 74</td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Configure H.323 Gateway, on page 76</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Many Cisco gateways can be deployed using any one of MGCP, SCCP, SIP, or H.323 as the gateway protocol. Review your gateway documentation to determine which protocols your gateway supports and which protocol is best for your deployment.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SCCP gateways can connect only to analog access or ISDN BRI connections.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Step 2**

Configure Clusterwide Call Classification for Gateway, on page 77

Optional. Configure a clusterwide service parameter to classify all calls coming from the
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway ports in your network to be internal (OnNet) or external (OffNet).</td>
<td>The call classification setting in the Port Configuration for individual gateway port interfaces overrides the clusterwide setting. However, the default setting for gateway ports is to use the setting from the clusterwide service parameter.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Block OffNet Gateway Transfers, on page 77</td>
<td>Optional. If you want to block Unified Communications Manager from transferring calls from one external (OffNet) gateway to another external gateway, configure the <strong>Block OffNet to Offnet Transfer</strong> service parameter. By default, this service parameter is configured to allow transfers from one external (OffNet) gateway to another.</td>
</tr>
</tbody>
</table>

### Configure MGCP Gateway

Perform the following tasks to configure a Cisco gateway to use an MGCP configuration.

**Before you begin**

**Gateway Setup Prerequisites, on page 65**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure MGCP (IOS) Gateway, on page 68</td>
<td>Add the gateway in Cisco Unified CM Administration and choose <strong>MGCP</strong> as the gateway protocol. Configure the gateway with the appropriate slots and vendor interface cards (VICs).</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure the gateway port interface. Select any of the following tasks, depending on the type of interface that you want to configure:</td>
<td>Configure the port connections for the devices that connect to the VICs that are installed on the gateway. Most VICs include multiple port connections and options so you may have to configure a few different port interface types.</td>
</tr>
<tr>
<td>• Configure Digital Access PRI Ports, on page 72</td>
<td>• Configure Digital Access PRI Ports, on page 72</td>
</tr>
</tbody>
</table>
### Purpose

Command or Action | Purpose
------------------|---------------------------------------------------
• Configure Digital Access T1 Ports for MGCP Gateway, on page 70  
• Configure FXS Ports, on page 69  
• Configure FXO Ports, on page 69  
• Configure BRI Ports, on page 72 | Tip After you configure a port interface, from the Related Links drop-down list box, select the Back to MGCP Configuration option to return to the Gateway Configuration window, where you can select and configure another port interface.

| Step 3 | Add Digital Access T1 Ports for MGCP Gateway, on page 71 | Optional. If you have configured a digital access T1 CAS port interface, add T1 CAS ports to the gateway. You can add ports on an individual basis or add a range of ports simultaneously.

| Step 4 | Reset Gateway, on page 73 | The configuration changes take effect after you reset the gateway.

### Related Topics

- Call Routing Overview, on page 125

### Configure MGCP (IOS) Gateway

Perform the following procedure to add and configure an MGCP (IOS) gateway on the Unified Communications Manager.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose Device > Gateway.
- **Step 2** Click Add New.
- **Step 3** From the Gateway Type drop-down list, select the gateway and click Next.
- **Step 4** From the Protocol drop-down list, choose MGCP and click Next.
- **Step 5** In the Configured Slots, VICs and Endpoints area, perform the following steps:
  a) From each Module drop-down list, select the slot that corresponds to the Network Interface Module hardware that is installed on the gateway.
  b) From each Subunit drop-down list, select the VIC that is installed on the gateway.
  c) Click Save.
    The Port icons appear. Each Port icon corresponds to an available port interface on the gateway. You can configure any port interface by clicking the corresponding port icon.
- **Step 6** Complete the remaining fields in the Gateway Configuration window. For more information on the fields, see the system Online Help.
- **Step 7** Click Save.
Configure FXS Ports

Configure Foreign Exchange Station (FXS) ports on an MGCP gateway. You can use FXS ports to connect the gateway to a Plain Old Telephone Service (POTS) legacy phone or to another legacy device such as a fax machine, speakerphone, legacy voice-messaging system, or Interactive Voice Response (IVR).

Before you begin

You must add a gateway before configuring ports.

Procedure

Step 1  In the Cisco Unified CM Administration, choose Device > Gateway.
Step 2  Click Find and select the gateway on which you want to configure FXS ports.
Step 3  In the Configured Slots, VICs, and Endpoints area, click the FXS Port icon for the port that you want to configure. The Port Selection area displays.
Step 4  From the Port Type drop-down list, choose the type of connection that you want to configure:
   - POTS—Select this option if you want to connect this port to a POTS device such as a legacy phone.
   - Ground Start—Select this option if you want to use ground as a start signaling to connect this port to an unattended legacy device such as a fax machine, legacy voice-messaging system, or IVR.
   - Loop Start—Select this option if you want to use a loop start signaling to connect this port to an unattended legacy device such as a fax machine, legacy voice-messaging system, or IVR.
Step 5  Click Next. The Port Configuration window displays the configuration for the port interface with an analog access as the device protocol.
Step 6  From the Device Pool drop-down list, select a device pool.
Step 7  Complete the remaining fields in the Port Configuration window. For more information on the fields and their configuration options, see the system Online Help.
Step 8  Click Save.
Step 9  (Optional) To configure more port interfaces on the MGCP IOS gateway, from the Related Links drop-down list, select Back to Gateway and click Go.

The Gateway Configuration window displays the available ports for the gateway.

When you have completed configuring more ports interfaces, see Reset Gateway, on page 73.

Configure FXO Ports

Configure Foreign Exchange Office (FXO) ports on an MGCP (IOS) gateway. You can use FXO ports to connect the gateway to the PSTN or a legacy PBX.
Unified Communications Manager assumes all loop-start trunks lack the positive disconnect supervision. Configure trunks with the positive disconnect supervision as ground start, so that the active calls can be maintained during a server failover.

**Before you begin**

Configure MGCP (IOS) Gateway, on page 68

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Gateway**.

**Step 2** Click **Find** and select the gateway for which you want to configure FXO ports.

**Step 3** From the **Configured Slots, VICs, and Endpoints** area, locate the **Module** and **Subunit** that contain the FXO port on which you want to set up an FXO port interface and click the **Port** icon for the port that you want to configure.

**Step 4** From the **Port Type** drop-down list, select either **Ground-Start** or **Loop-Start**.

*Note* If you are configuring the VIC-2 FXO port, you must select the same port type for both ports of the subunit module.

**Step 5** From the **Device Pool** drop-down list, select a device pool.

**Step 6** In the **Attendant DN** text box, enter the directory number to which you want to route all incoming calls from this port connection. For example, a zero or the directory number for an attendant.

**Step 7** Complete any remaining fields in the **Port Configuration** window. Refer to the online help for field descriptions.

**Step 8** Click **Save**.

**Step 9** (Optional) To configure more port interfaces on the MGCP IOS gateway, from the **Related Links** drop-down list, select **Back to Gateway** and click **Go**.

The **Gateway Configuration** window displays the available ports for the gateway.

When you have completed configuring more ports interfaces, see **Reset Gateway, on page 73**.

---

**Configure Digital Access T1 Ports for MGCP Gateway**

Configure the port interface for digital access T1 CAS ports on an MGCP (IOS) gateway.

**Before you begin**

Configure MGCP (IOS) Gateway, on page 68

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Gateway**.
Step 2  Click **Find** and select the gateway on which you want to configure a T1 port.

Step 3  In the **Configured Slots, VICs and Endpoints** area, locate the Module and Subunit on which you want to set up a Digital Access T1 (T1-CAS) port and click the corresponding **Port** icon.

Step 4  From the **Device Protocol** drop-down list, choose **Digital Access T1** and click **Next**.

Step 5  Enter the appropriate gateway configuration settings.

For more information on the fields and their configuration options, see the system Online Help.

Step 6  Click **Save**.

For more information on adding ports to the Digital Access T1 CAS port interface, see **Add Digital Access T1 Ports for MGCP Gateway**, on page 71.

---

Add Digital Access T1 Ports for MGCP Gateway

Add and configure T1 CAS ports to a T1 Digital Access port interface for an MGCP gateway. You can add and configure up to 24 T1 CAS ports. You can also add ports on an individual basis or add and configure a range of ports simultaneously. If you enter a range of ports, Unified Communications Manager applies the configuration to the entire range of ports.

**Before you begin**

Configure Digital Access T1 Ports for MGCP Gateway, on page 70

**Procedure**

---

Step 1  In Cisco Unified CM Administration, choose **Device > Gateway**.

Step 2  Click **Find** and select the gateway that contains the T1 CAS port interface.

Step 3  Click **Add a New Port**.

Step 4  From the **Port Type** drop-down list, select the type of port that you want to add and click **Next**.

Step 5  Enter port numbers in the **Beginning Port Number** and **Ending Port Number** fields to specify the range of ports that you want to add and configure.

For example, enter 1 and 10 to add ports 1 through 10 to the port interface simultaneously.

Step 6  From the **Port Direction** drop-down list, configure the direction of calls passing through this port:

- **Bothways**—Select this option if the port allows both inbound and outbound calls.
- **Inbound**—Select this option if the port allows inbound calls only.
- **Outbound**—Select this option if the port allows outbound calls only.

Step 7  For EANDM ports, from the **Calling Party Selection** drop-down list, choose how you want the calling number to display for outbound calls from the device that is attached to this port:

- **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 directory number of the first redirecting device with an external phone mask applied.
Configure Digital Access PRI Ports

Configure the PRI port interface for an MGCP (IOS) gateway.

Before you begin
Configure MGCP (IOS) Gateway, on page 68

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Gateway.
Step 2 Click Find and select the gateway on which you want to configure PRI ports.
Step 3 In the Configured Slots, VICs, and Endpoints area, locate the Module and Subunit that contains the BRI port that you want to configure and click the Port icon that corresponds to the BRI port that you want to configure.
   The Gateway Configuration window displays the BRI port interface.
Step 4 From the Device Pool drop-down list, select a device pool.
Step 5 Complete the remaining fields in the Gateway Configuration window. Refer to the online help for field descriptions.
Step 6 Click Save.
Step 7 (Optional) If you want to configure more port interfaces for the gateway, from the Related Links drop-down list, choose Back to MGCP Configuration and click Go.
   The Gateway Configuration window displays the available port interfaces for the gateway.
   When you have completed configuring more ports interfaces, see Reset Gateway, on page 73.

Configure BRI Ports

Configure a BRI port interface for an MGCP (IOS) gateway.
Before you begin

Configure MGCP (IOS) Gateway, on page 68

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Gateway.
Step 2  Click Find and select the gateway on which you want to configure BRI ports.
Step 3  In the Configured Slots, VICs, and Endpoints section, locate the subunit that uses BRI ports and click the Port icon for the port that you want to configure. The Gateway Configuration window displays the information for the BRI port interface.
Step 4  From the Device Pool drop-down list, select a device pool.
Step 5  Enter the appropriate Gateway Information and Port Information settings. For more information on the fields and their configuration options, see the system Online Help.
Step 6  Click Save.
Step 7  (Optional) If you want to configure more port interfaces for the gateway, from the Related Links drop-down list, choose Back to MGCP Configuration and click Go.

The Gateway Configuration window displays the available port interfaces for the MGCP gateway.

When you have completed configuring more ports interfaces, see Reset Gateway, on page 73.

Reset Gateway

Most gateways need to be reset for configuration changes to take effect. We recommend that you complete all necessary gateway configuration before performing a reset.

Note

Resetting an H.323 gateway only reinitializes the configuration that Unified Communications Manager loaded and does not physically restart or reset the gateway.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Gateway.
Step 2  Click Find and select the gateway.
Step 3  Click the check box beside the gateway that you want to reset and click Reset Selected. The Device Reset dialog box appears. Do one of the following actions:
Step 4  Click Reset.
Configure SCCP Gateway

You can configure a Cisco gateway to use SCCP as the gateway protocol. You can use this deployment option to connect Unified Communications Manager to analog access devices or ISDN BRI devices using FXS or BRI ports. You cannot connect an SCCP gateway to digital access T1 or E1 trunks.

Procedure

| Step 1 | From Cisco Unified CM Administration, choose Device > Gateway. |
| Step 2 | Click Add New. |
| Step 3 | From the Gateway Type drop-down list, choose a gateway that uses SCCP and click Next. |
| Step 4 | From the Protocol drop-down list, choose SCCP. |
| Step 5 | In the Configured Slots, VICs and Subunits section, perform the following steps: |
| a) | For each Module drop-down list, select the slot that corresponds to the Network Interface Module hardware that is installed on the gateway. |
| b) | For each Subunit, select the VIC that is installed on the gateway. |
| Step 6 | Complete the remaining fields in the Gateway Configuration window. |
| Step 7 | For more information on the fields and their configuration options, see the system Online Help. |
| Step 8 | Click Save. |

Configure SIP Gateway

Perform the following tasks to configure a SIP gateway in Unified Communications Manager. Many Cisco gateways and third-party gateways can be configured to use SIP. Unified Communications Manager does not contain a gateway device type for SIP gateways.

Before you begin

You must install the gateway hardware in your network and configure the IOS software on the gateway before you add the gateway in Unified Communications Manager.

Procedure

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Configure SIP Profile

Configure a SIP profile for your SIP gateway connection.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
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</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; SIP Profile.</td>
<td></td>
</tr>
</tbody>
</table>
| Step 2 | Perform either of the following steps:  
   - Click Add New to create a new profile.  
   - Click Find to select an existing SIP profile. | |
| Step 3 | Complete the fields in the SIP Profile Configuration window.  
For more information on the fields and their configuration options, see the system Online Help. | |
| Step 4 | Click Save. | |

Configure SIP Trunk Security Profile.

Configure a SIP trunk security profile with security settings for a trunk that connects to a SIP gateway.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
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<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified CM Administration, choose System &gt; Security &gt; SIP Trunk Security Profile.</td>
<td></td>
</tr>
</tbody>
</table>
| Step 2 | Perform either of the following steps:  
   a) Click Find to select an existing profile.  
   b) Click Add New to create a new profile. | |
| Step 3 | Complete the fields in the SIP Trunk Security Profile Configuration window.  
For more information on the fields and their configuration options, see the system Online Help. | |
| Step 4 | Click Save. | |
Configure SIP Trunk for SIP Gateway

Configure a SIP trunk to connect Unified Communications Manager to a Cisco or third party gateway that uses SIP. Under this configuration, do not enter the gateway as a device in the Gateway Configuration window.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Trunk.
Step 2  Click Add New to set up a new SIP trunk.
Step 3  From the Trunk Type drop-down list choose SIP Trunk.
Step 4  From the Protocol drop-down list, choose None.
Step 5  In the Destination Address field of the SIP Information pane, enter an IP address, fully qualified domain name, or DNS SRV record for the SIP gateway.
Step 6  From the SIP Trunk Security Profile drop-down list, choose the SIP trunk security profile that you configured for this gateway.
Step 7  From the SIP Profile drop-down list box, choose the SIP profile that you configured for this gateway.
Step 8  Complete the fields in the SIP Trunk Configuration window. Refer to the online help for field descriptions.
Step 9  Click Save.

Configure H.323 Gateway

Configure an H.323 gateway in Unified Communications Manager for a non-gatekeeper H.323 deployment.

Note

If your deployment includes H.323 gatekeepers, you can also add an H.323 gateway by setting up a gatekeeper-controlled H.225 trunk. This scenario is not documented in this guide because gatekeeper usage has been in steady decline recent years. If you want to configure gatekeepers and H.225 gatekeeper-controlled trunks, refer to the Cisco Unified Communications Manager Administration Guide, Release 10.0(1).

Note

When a gateway is registered with Unified Communications Manager, the registration status may display in Unified Communications Manager Administration as unknown.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Gateway.
Step 2  Click Add New.
Step 3  From the Gateway Type drop-down list, choose H.323 Gateway.
Step 4  In the Device Name field, enter the IP address or hostname of the gateway.
Step 5  If you want to use H.235 to configure a secure channel, check the H.235 Data Passthrough check box.
Step 6 Configure the fields in the Gateway Configuration window.
For more information on the fields and their configuration options, see the system Online Help.

Step 7 Click Save.

Step 8 Click Reset to reset the gateway and apply the changes.
Most gateways need to be reset for configuration changes to take affect. We recommend that you complete all necessary gateway configuration before performing a reset.

---

**Configure Clusterwide Call Classification for Gateway**

Configure the Call Classification setting for your network gateways. This setting determines whether the system considers the gateways in the network to be internal (OnNet) or external (OffNet).

The Call Classification field also appears in the configuration window for individual gateway port interfaces. By default, each gateway port interface is configured to use the setting from the clusterwide service parameter. However, if Call Classification on a port is configured differently from the clusterwide service parameter, the setting on that port overrides the service parameter setting.

**Procedure**

Step 1 From Cisco Unified CM Administration, choose System > Service Parameters.

Step 2 From the Server drop-down list, choose the server on which the Cisco CallManager service is running.

Step 3 From the Service drop-down list, choose Cisco CallManager.

Step 4 Under Clusterwide Parameters (Device - General), configure one of the following values for the Call Classification service parameter.

- **OnNet**—Calls from this gateway are classified as originating from inside the company network.
- **OffNet**—Calls from this gateway are classified as originating from outside the company network.

Step 5 Click Save.

---

**Block OffNet Gateway Transfers**

Use this procedure if you want to configure the system to block calls that are transferred from one external (OffNet) gateway to another external (OffNet) gateway. By default, the system allows transfers from one external gateway to another external gateway.

The setting that determines whether a gateway is external (OffNet) or internal (OnNet) is determined by the Call Classification setting. It is configured using a clusterwide service parameter, or by configuring any of the following port interfaces:

- MGCP T1/E1 port interfaces
- MGCP FXO port interface
- H.323 gateways
Block OffNet Gateway Transfers

- SIP trunks

Procedure

**Step 1**
From Cisco Unified CM Administration, choose System > Service Parameters.

**Step 2**
From the Server drop-down list, choose the server on which the Cisco CallManager service is running.

**Step 3**
From the Service drop-down list, choose Cisco CallManager.

**Step 4**
Configure a setting for the **Block OffNet to Offnet Transfer** service parameter:

- **True**—Select this option to cancel transfers between two external (OffNet) gateways.
- **False**—Select this option to allow transfers between two external (OffNet) gateways. This is the default option.

**Step 5**
Click Save.

**Note**
You can also classify calls through a gateway as OnNet or OffNet by associating the gateway to a route pattern and configure **Call Classification** in the **Route Pattern Configuration** window.
CHAPTER 10

Configure SIP Normalization and Transparency

• SIP Normalization and Transparency Overview, on page 79
• SIP Normalization and Transparency Prerequisites, on page 80
• SIP Normalization and Transparency Configuration Task Flow, on page 80

SIP Normalization and Transparency Overview

SIP normalization and transparency is an optional feature that handles SIP interoperability issues between Unified Communications Manager and endpoints, service providers, PBXs, or gateways that implement SIP differently. To configure SIP normalization and transparency, apply a customized LUA script to a SIP trunk or SIP line. Unified Communications Manager applies the script to the SIP messaging that passes through the SIP trunks or SIP lines.

Upon installation, Unified Communications Manager contains default normalization and transparency scripts that you can assign to the SIP trunks and SIP profiles in your system. You can also create and import your own customized scripts.

SIP Normalization

SIP normalization scripts modify incoming and outgoing SIP messages. For example, if you are interoperating Unified Communications Manager with a Cisco TelePresence Video Communications Server, apply the vcs-interop script to the SIP trunk that connects the two. The script resolves the differences in the SIP messaging so that the two products can communicate.

You can apply a normalization script to any SIP trunk connection, regardless of which protocol is being used by the endpoint that connects to that SIP trunk.

SIP Transparency

SIP transparency scripts enable Unified Communications Manager to transparently pass SIP information, such as proprietary headers, from one call leg to the other. For transparency to work, both call legs must be SIP.

Another feature of SIP transparency is REFER transparency, which allows Unified Communications Manager to pass on REFER requests without acting on them. You can use REFER transparency in call center environments where a centralized agent may answer a call and then transfer the call to an agent who resides in the same geographical area as the caller. REFER transparency allows the centralized Unified Communications Manager to drop the call and shift call control to the new agent.
Default Scripts for SIP Normalization and Transparency

Upon installation, Cisco Unified Communications Manager contains the following default scripts for SIP Normalization and Transparency. You can apply these scripts to a SIP trunk or SIP profile, but you cannot edit these scripts. If none of these scripts meet your needs, you can create your own scripts:

- **HCS-PCV-PAI passthrough** — Provides Cisco HCS platform integration with Enterprise IMS.
- **cisco-telepresence-conductor-interop** — Provides interoperability for endpoints registered to TelePresence Conductor.
- **cisco-telepresence-mcu-ts-direct-interop** — Provides interoperability between Cisco Unified Communications Manager and either Cisco TelePresence MCU or Cisco TelePresence Server.
- **diversion-counter** — Provides capability to adjust the diversion counter.
- **refer-passthrough** — Removes Cisco Unified Communications Manager from the call due to a blind transfer between SIP trunks.
- **vcs-interop** — Provides interoperability for endpoints that are registered to the Cisco TelePresence Video Communications Server.

SIP Normalization and Transparency Prerequisites

- Cisco Unified Communications Manager provides default scripts for SIP Normalization and Transparency. Make sure to review the existing scripts and system settings to verify whether they meet your needs. For information on the available default scripts, see Default Scripts for SIP Normalization and Transparency, on page 80.

SIP Normalization and Transparency Configuration Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
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</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Create New SIP Normalization and Transparency Scripts, on page 81</td>
<td>Optional. If none of the preinstalled scripts meet your needs, use this procedure to configure a customized script. You can create your new script in the SIP Normalization Script</td>
</tr>
</tbody>
</table>
### Create New SIP Normalization and Transparency Scripts

If the default normalization and transparency scripts do not meet your needs, use this procedure to create a new LUA script. You can either write the new script in Cisco Unified Communications Manager or import a file into the system.

#### Tip

If the script that you want to create closely resembles a default script, open the default script in the **SIP Normalization Script Configuration** window and copy the **Contents** text box. Create a new script and paste the contents into the **Contents** text box. You can then edit the content in the new script.

#### Note

The memory utilization of the **SIP Normalization Script** is based on each trunk and not on each script.

#### Procedure

1. **Step 1**
   - In Cisco Unified CM Administration, choose **Device > Device Settings > SIP Normalization Script**.
2. **Step 2**
   - Click **Add New**.
   - The SIP Normalization Script Configuration window appears.
3. **Step 3**
   - Enter a **Name** and **Description** for your script.
4. **Step 4**
   - If you are writing a new script, edit the script in the **Contents** text box.
5. **Step 5**
   - Optional. If you have an external file that you want to import, do the following:
     a) Click **Import File**.
     b) **Browse** to locate the file and select the file.
     c) Click **Import File**.

     The **SIP Normalization Script Configuration** window displays the contents of the imported file in the **Contents** text box.
Apply Normalization or Transparency Script to SIP Trunk

Use this procedure to apply a SIP normalization or transparency script to a SIP trunk. Cisco Unified Communications Manager applies the script to all SIP messaging that passes through the trunk.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Trunk.
Step 2  Click Find and select the trunk to which you want to apply a script.
Step 3  From the Normalization Script drop-down list, choose the script that you want to apply to the trunk.
Step 4  (Optional) If you want to normalize specific parameters within the SIP messaging, do the following:
   a) Enter the Parameter Name that you want to normalize, and the Parameter Value for the value that you want to apply to the parameter. For example, you could enter a Location parameter and North Carolina as the value.
   b) To add additional parameters, click the (+) to create additional lines where you can enter additional parameters and values.
Step 5  (Optional) If you want to produce SDI traces against the script, check the Enable Trace check box.
   Note    Cisco recommends that you enable tracing while debugging your scripts.
Step 6  Click Save.

Apply Normalization or Transparency Script to SIP Devices

You can apply a customized SIP Normalization and Transparency script, or a customized SDP Transparency Profile to a SIP phone by applying the script to the SIP Profile that is used by that device.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.
Step 2  Click Find and select the SIP profile to which you want to apply a script.
Step 3  In the SDP Information area, from the SDP Transparency Profile drop-down list, choose a profile.

Step 4  From the Normalization Script drop-down list, choose the script that you want to apply to the trunk.

Step 5  (Optional) If you want to normalize specific parameters within the SIP messaging, do the following:
   a) Enter the Parameter Name that you want to normalize, and the Parameter Value for the value that you want to apply to the parameter. For example, you could enter a Location parameter and North Carolina as the value.
   b) To add additional parameters, click the (+) to create additional lines where you can enter additional parameters and values.

Step 6  (Optional) If you want to produce SDI traces against the script, check the Enable Trace check box.

Note  Cisco recommends that you enable tracing while debugging your scripts.

Step 7  Click Save.
Apply Normalization or Transparency to SIP Devices
CHAPTER 11

Configure SDP Transparency Profiles

- SDP Transparency Profile Overview, on page 85
- SDP Transparency Profile Restrictions, on page 85
- SDP Transparency Profile Prerequisites, on page 86
- Configure SDP Transparency Profile, on page 86

SDP Transparency Profile Overview

SDP Transparency Profiles contain a set of rules for declarative SDP attributes that allow the system to pass through declarative attributes that are not natively supported by Unified Communications Manager from the ingress to the egress call leg. Without an SDP transparency profile, Unified Communications Manager drops non-supported SDP attributes.

You can configure SDP transparency profiles with multiple rules and apply them to SIP devices via the SIP profile. In order for the SDP transparency profile to be applied, both call legs must be SIP. You can configure the following types of rules for SDP attributes:

- Property—If a rule is configured for a property attribute, Unified Communications Manager passes through the SDP attribute unless the attribute has a value.
- Any Value—If a rule is configured for any value, the SDP attribute gets passed through so long as it has a value that consists of at least one non-white space character.
- Value From List—If a rule is configured using this option, the SDP attribute gets passed through so long as it matches one of the specified values. You can configure up to five possible values

SDP Transparency Profile Restrictions

The following restrictions apply to SDP transparency profiles. If any of these situations occur on the egress call leg, Cisco Unified Communications Manager will not pass through the declarative SDP attribute:

- One or more Media Termination Points (MTPs) or Trusted Relay Points (TRPs) that do not support passthrough are allocated
- The Media Termination Point Required check box is checked for the SIP trunk
- A transcoder is being used
- RSVP is being used
• The ingress call leg is using Delayed Offer while the egress call leg is using Early Offer
• The media line has been rejected (port=0)
• Either call leg is using a protocol other than SIP

SDP Transparency Profile Prerequisites

If you plan to deploy any third-party SIP products, make sure that you understand how the products implement the Session Description Protocol (SDP).

Configure SDP Transparency Profile

Configure a customized SDP Transparency Profile with a set of rules for declarative SDP attributes that are not natively supported by Cisco Unified Communications Manager.

Procedure

| Step 1 | From Cisco Unified CM Administration, choose Device > Device Settings > SDP Transparency Profile. |
| Step 2 | Click Add New. |
| Step 3 | Enter a Name and Description. |
| Step 4 | In the Attribute Information pane, create the rules for the SDP attributes that you want to pass through: |
| | • To pass through a property attribute, enter the attribute in the Name text box (for example, a=recvonly) and from the Type drop-down list, select Property. |
| | • To pass through a value attribute, enter the attribute in the Name text box (for example, a=rtpmap), and select Any Value from the Type drop-down list box. |
| | • To pass through a value attribute with any of up to five values, enter the attribute in the Name field (for example, a=rtpmap) and select Any Value from the Type drop-down list. In the resulting Value text box, enter the value of the attribute. You can click + to add up to five possible values for this attribute. |
| Step 5 | Click the (+) to create new lines where you can enter additional SDP attributes for this transparency profile. |
| Step 6 | Click Save. |

Note You must apply this profile to a SIP Profile so that the devices that use the SIP Profile can use the SDP Transparency Profile.
Configure SIP Profiles

- SIP Profile Overview, on page 87
- Configure SIP Profiles, on page 87

SIP Profile Overview

A SIP Profile is a template that comprises common SIP settings. If you are deploying SIP devices or SIP trunks in your network, you can apply common SIP settings to groups of devices through the SIP Profile. You can configure multiple profiles for groups of SIP endpoints. You can choose from a variety of default SIP Profiles or create your own.

Without the SIP Profile, you would have to configure SIP settings individually for every SIP trunk and SIP device in your network. However, you can use the SIP profile to assign many SIP settings, such as the following:

- MTP Telephony Payload Types
- SIP header details
- Timers and counters for SIP messages
- SDP transparency profiles for SDP interoperability
- SIP Normalization and Transparency scripts for SIP lines
- SIP OPTIONS settings
- SIP Early Offer support
- Call Pickup URIs

Configure SIP Profiles

Use this procedure to configure a SIP profile with common SIP settings that you can assign to SIP devices and trunks that use this profile.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.
Step 2 Perform one of the following steps:
   - Click **Find** and select the SIP profile to edit an existing profile.
   - Click **Add New** to create a new profile.

Step 3 If you want your SIP phones and trunks to support IPv4 and IPv6 stacks, check the **Enable ANAT** check box.

Step 4 If you want to assign an SDP transparency profile to resolve SDP interoperability, from the **SDP Transparency Profile** drop-down list.

Step 5 If you want to assign a normalization or transparency script to resolve SIP interoperability issues, from the **Normalization Script** drop-down list, select the script.

Step 6 (Optional) Check the **Send ILS Learned Destination Route String** check box for Global Dial Plan Replication deployments where you may need to route calls across a Cisco Unified Border Element.

Step 7 Complete the remaining fields in the **SIP Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 8 Click **Save**.
CHAPTER 13

Configure Dual Stack IPv6

• Dual Stack Addressing Overview, on page 89
• Dual Stack IPv6 Prerequisites, on page 90
• Dual Stack IPv6 Configuration Task Flow, on page 90

Dual Stack Addressing Overview

If your SIP deployment requires IPv6 addressing, you can configure Cisco Unified Communications Manager to support dual stack IPv4 and IPv6 addressing. By default, Cisco Unified Communications Manager is enabled for IPv4 addressing. At the system level, you cannot configure Cisco Unified Communications Manager to support IPv6 only because the system must still interact with devices that support IPv4 only. However, if you require IPv6 addressing, you can configure dual stack trunks and devices.

Dual Stack IPv6 at the System Level

When Cisco Unified Communications Manager is configured for dual stack addressing the system can set up calls for the following scenarios:

• All devices in the call support IPv4 only
• All devices in the call support IPv6 only
• All devices in the call run in dual-stack mode—In this scenario, the system determines the IP address type by the configuration for the IP Addressing Mode Preference for Signaling setting for signaling events and the IP Addressing Mode Preference for Media enterprise parameter for media events.
• One device supports IPv4 only and the other supports IPv6 only—In this scenario, Cisco Unified Communications Manager inserts an MTP into the call path to translate the signaling between the two addressing types.

Cisco Unified Communications Manager supports IPv6 addresses for SIP environments only. For H.323 deployments, the system inserts an MTP into the call path so that the IPv4 and IPv6 devices can communicate.

Dual Stack IPv6 for Devices

At the device level you can configure many devices and media resources such as phones, gateways, conference bridges can be configured to use IPv4 addressing only, IPv6 addressing only, or dual stack. You can configure the preferred addressing method both for signaling and for media events.
For SIP devices, you can also configure the Alternate Network Address Types (ANAT) feature allowing a registered SIP device to have both an IPv4 and IPv6 address at the same time. The device can communicate using either address type, thereby allowing the device to interoperate seamlessly in both IPv4 and IPv6 networks. You can enable ANAT for SIP devices by enabling ANAT in the SIP Profile that is assigned to the device.

Dual Stack IPv6 Prerequisites

Before you configure Cisco Unified Communications Manager with dual stack IPv6 support, you must configure the following network servers and devices to support IPv6. For details, refer to your device user documentation:

- Provision a DHCP and DNS server with IPv6 support. The Cisco Network Registrar server supports IPv6 for DHCP and DNS.
- Configure the IOS for network devices such as gateways, routers, and MTPs with IPv6 support.
- Configure your TFTP server to run IPv6.

Dual Stack IPv6 Configuration Task Flow

Complete the following tasks to configure the system for dual stack IPv6.

### Procedure

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<th>Step</th>
<th>Command or Action</th>
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<td>Configure the operating system with support for IPv6 addresses.</td>
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<tr>
<td><strong>Step 2</strong></td>
<td>Configure Server for IPv6, on page 91</td>
<td>Configure the servers in your cluster with IPv6 addresses.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Enable IPv6, on page 92</td>
<td>Configure enterprise parameters that enable the system for IPv6.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Perform any of the following:  - Configure IP Addressing Preference for Cluster, on page 93  - Configure IP Addressing Preferences for Devices, on page 93</td>
<td>You can configure an enterprise parameter to assign a clusterwide IP Addressing preference. If you want to assign different preferences for different groups of endpoints, configure the addressing preference within a Common Device Configuration. Configure cluster settings for which IP addressing method is preferred.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Restart Services, on page 94</td>
<td>Restart the following network services:  - Cisco CallManager  - Cisco CTIManager</td>
</tr>
</tbody>
</table>
Configure IPv6 in Operating System

Use this procedure to set up Ethernet IPv6 in Cisco Unified OS Administration.

Procedure

Step 1 From Cisco Unified OS Administration, choose **Settings > IPv6 > Ethernet**.
Step 2 Check the **Enable IPv6** check box.
Step 3 From the **Address Source** drop-down list box, configure how the system acquires the IPv6 address:
   - **Router Advertisement**—The system uses stateless autoconfiguration to acquire an IPv6 address.
   - **DHCP**—The system acquires an IPv6 address from a DHCP server.
   - **Manual Entry**—Choose this option if you want to enter the IPv6 address manually.
Step 4 If you have configured Manual Entry as the means of acquiring an IPv6 address, complete the following fields:
   - Enter an **IPv6 Mask**. For example, `64`.
Step 5 Check the **Update with Reboot** check box to ensure that the system reboots after you save.
Step 6 Click **Save**.

Configure Server for IPv6

Configure the servers in your cluster with IPv6 addresses.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System > Server**.
Step 2 In the **IPv6 Name** field, enter one of the following values:
   - If you have DNS configured, and your DNS server supports IPv6, enter the server hostname.
   - Otherwise, enter the non-link local IPv6 address.
Enable IPv6

If you want to set up IPv6 support in your system, you must enable the system to support IPv6 devices.

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Enterprise Parameters.
Step 2  Set the value of the Enable IPv6 enterprise parameter to True.
Step 3  Click Save.

What to do next

Configure IP addressing preferences for the devices in your cluster. You can apply settings via a clusterwide enterprise parameter or you can use a Common Device Configuration to apply settings to a group of devices that uses that configuration:

- Configure IP Addressing Preference for Cluster, on page 93
- Configure IP Addressing Preferences for Devices, on page 93

Configure IP Addressing Preference

You can configure the IP Addressing preference for individual trunks or SIP devices by configuring a Common Device Configuration that includes the IP address preferences and then applying that configuration to a trunk or device.

Procedure

<table>
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<tr>
<th>Command or Action</th>
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<tbody>
<tr>
<td><strong>Step 1</strong> Configure IP Addressing Preference for Cluster, on page 93</td>
<td>Configure the IP Address preference at the system level using clusterwide enterprise parameters. The setting applies to all SIP devices and trunks in the cluster unless an overriding Common Device Configuration is applied to a trunk or device.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure IP Addressing Preferences for Devices, on page 93</td>
<td>Configure a Common Device Configuration with IP addressing preferences. You can apply the configuration to dual stack devices such SIP trunks, SIP phones, conference bridges, and transcoders.</td>
</tr>
</tbody>
</table>
## Configure IP Addressing Preference for Cluster

Use this procedure to use enterprise parameters to configure clusterwide IP addressing preferences for dual stack IPv6. The system applies these settings to all SIP trunks and devices unless an overriding Common Device Configuration is applied to a specific trunk or device.

### Notes
The IP address preferences in a Common Device Configuration override the clusterwide enterprise parameter settings for the devices that use that Common Device Configuration.

### Procedure

1. From Cisco Unified CM Administration, choose **System > Enterprise Parameters**.
2. Set the value of the **IP Addressing Mode Preference for Media** enterprise parameter to IPv4 or IPv6.
3. Set the value of the **IP Addressing Mode Preference for Signaling** enterprise parameter to IPv4 or IPv6.
4. Click **Save**.

## Configure IP Addressing Preferences for Devices

You can configure IP addressing preferences for individual devices by configuring a Common Device Configuration with the preference settings. You can apply the Common Device Configuration to SIP and SCCP devices that support IPv6 addressing such as trunks, phones, conferences bridges, and transcoders.

### Notes
The IP address preferences in a Common Device Configuration override the clusterwide enterprise parameter settings for the devices that use that Common Device Configuration.

### Procedure

1. From Cisco Unified CM Administration, choose **Device > Device Settings > Common Device Configuration**.
2. Click **Add New**.
3. For SIP trunks, SIP Phones or SCCP phones, choose a value for the **IP Addressing Mode** drop-down list:
   - IPv4 Only—The device uses only an IPv4 address for media and signaling.
• **IPv6 Only**—The device uses only an IPv6 address for media and signaling.

• **IPv4 and IPv6 (Default)**—The device is a dual-stack device and uses whichever IP address type is available. If both IP address types are configured on the device, for signaling the device uses the [IP Addressing Mode Preference for Signaling](#) setting and for media the device uses the [IP Addressing Mode Preference for Media](#) enterprise parameter setting.

**Step 4** If you configure IPv6 in your previous step, then configure an IP addressing preference for the [IP Addressing Mode for Signaling](#) drop-down list:

- **IPv4**—The dual stack device prefers IPv4 address for signaling.
- **IPv6**—The dual stack device prefers IPv6 address for signaling.
- **Use System Default**—The device uses the setting for the [IP Addressing Mode Preference for Signaling](#) enterprise parameter.

**Step 5** Configure the remaining fields in the **Common Device Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 6** Click **Save**.

---

**What to do next**

If your IPv6 configuration is complete, [Restart Services](#) on page 94.

### Restart Services

After configuring your system for IPv6, restart essential services.

**Procedure**

**Step 1** Log into Cisco Unified Serviceability and choose **Tools > Control Center - Feature Services**.

**Step 2** Check the check box corresponding to each of the following services:

- Cisco CallManager
- Cisco CTIManager
- Cisco Certificate Authority Proxy Function
- Cisco IP Voice Media Streaming App

**Step 3** Click **Restart**.

**Step 4** Click **OK**.
Configure SIP Trunks

• SIP Trunk Overview, on page 95
• SIP Trunk Configuration Prerequisites, on page 97
• SIP Trunk Configuration Task Flow, on page 97

SIP Trunk Overview

If you are deploying SIP for call control signaling, configure SIP trunks that connect Cisco Unified Communications Manager to external devices such as SIP gateways, SIP Proxy Servers, Unified Communications applications, remote clusters, or a Session Management Edition.

Within Cisco Unified CM Administration, the SIP Trunk Configuration window contains the SIP signaling configurations that Cisco Unified Communications Manager uses to manage SIP calls.

You can assign up to 16 different destination addresses for a SIP trunk, using IPv4 or IPv6 addressing, fully qualified domain names, or you can use a single DNS SRV record.

You can configure the following features on SIP trunks:

• Line and Name Identification Services
• Delayed Offer, Early Offer and Best Effort Early Offer
• Signaling encryption and authentication
• Media encryption with SRTP
• IPv6 dual stack support
• Video
• Presentation sharing with BFCP
• Far end camera control
• DTMF relay
• Calling party normalization
• URI dialing
• Q.SIG support
- T.38 fax support
- SIP OPTIONS
- Choice of DTMF signaling

**Note**
When Q.SIG is enabled in Small-scale IP telephony (SIPT) from Cluster A to Cluster B, and if "INVITE" is received with anonymous or any text, then the Cisco Unified Communications Manager does not encode it to Q.SIG data. When you decode the same in the leaf cluster, it displays empty and empty number is forwarded.

**Note**
When Q.SIG is enabled, URI dialing does not respond as expected and if Q.SIG is disabled, then the Cisco Call Back does not respond between two clusters.

**IPv6 Dual Stack Support**
You can also configure your SIP trunks with IPv6 dual stack support by configuring the IP Addressing Mode in a Common Device Configuration and then applying that configuration to the SIP trunk.

**Note**
You can also configure IPv6 clusterwide via a clusterwide service parameter. However, the Common Device Configuration setting overrides the clusterwide defaults.

**Secure SIP Trunks**
You can also configure your trunks with security such as digest authentication and signaling and media encryption by configuring a SIP trunk security profile that includes security features such as digest authentication and TLS signaling and associate that profile to the SIP trunks in your network. For the trunk to allow encrypted or encrypt call media, you must also configure the trunk to allow SRTP media.

**SIP Trunk Security Profile Overview**
You must assign a SIP trunk security profile to each SIP trunk in your network. By default, Cisco Unified Communications Manager applies a predefined, nonsecure SIP trunk security profile for autoregistration to all SIP trunks.

The SIP trunk security profile allows you to configure security settings such as digest authentication and TLS signaling encryption for the SIP trunks in your network. When you configure a SIP trunk security profile, and then assign that profile to a SIP trunk, the security settings from the profile get applied to the trunk.

You can configure multiple SIP trunk security profiles to cover the different security requirements that you have for different sets of SIP trunks in your network.

**Note**
To configure your network with security, you must also set up a CTL client and configure IPSec. For details, see the *Security Guide for Cisco Unified Communications Manager*. 

---

**Enable Inbound and Outbound Calling**

---

---

---
SIP Trunk Configuration Prerequisites

Before you configure your SIP trunks, do the following:

- Plan your network topology so that you understand your trunk connections.
- Make sure that you understand the devices to which you want to connect your trunks and how those devices implement SIP. If those devices implement SIP, you may need to apply a SIP normalization script.
- Configure SIP profiles for your trunks.

In addition, configure the following before you configure your SIP trunks:

- SIP Normalization and Transparency Configuration Task Flow, on page 80
- Configure SIP Profiles, on page 87

SIP Trunk Configuration Task Flow

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure SIP Trunk Security Profile, on page 98</td>
<td>Configure SIP trunk security profiles with any security settings that you want to apply to your SIP trunks. For example, you can configure digest authentication, device security mode, and TLS encryption for SIP signaling. If you don't configure SIP trunk security profiles, by default, Cisco Unified Communications Manager applies a nonsecure sip trunk security profile.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Common Device Configuration, on page 98</td>
<td>Set up a Common Device Configuration for the trunk. For dual-stack trunks, configure the IP addressing preference.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure SIP Trunks, on page 99</td>
<td>Configure the SIP trunks in your network. In the Trunk Configuration window, configure the SIP settings for your trunks. Assign a SIP profile, SIP trunk security profile, and a Common Device Configuration to your SIP trunk. In addition, assign any SIP normalization or transparency scripts that your trunk connection requires. For example, if your SIP trunk connects to a Cisco TelePresence VCS, you must assign the vcs-interop script to the SIP trunk.</td>
</tr>
</tbody>
</table>
Configure SIP Trunk Security Profile

Configure a SIP Trunk Security Profile with security settings such as digest authentication or TLS signaling encryption. When you assign the profile to a SIP trunk, the trunk takes on the settings of the security profile.

**Note**

If you don't assign a SIP trunk security profile to your SIP trunks, Cisco Unified Communications Manager assigns a nonsecure profile by default.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose *System > Security > SIP Trunk Security Profile.*

**Step 2** Click *Add New.*

**Step 3** To enable SIP signaling encryption with TLS, perform the following:
- a) From the *Device Security Mode* drop-down list, select *Encrypted.*
- b) From the *Incoming Transport Type* and *Outgoing Transport Type* drop-down lists, choose *TLS.*
- c) For device authentication, in the *X.509 Subject Name* field, enter the subject name of the X.509 certificate.
- d) In the *Incoming Port* field, enter the port on which you want to receive TLS requests. The default for TLS is 5061.

**Step 4** To enable digest authentication, do the following
- a) Check the *Enable Digest Authentication* check box
- b) Enter a *Nonce Validity Timer* value to indicate the number of seconds that must pass before the system generates a new nonce. The default is 600 (10 minutes).
- c) To enable digest authentication for applications, check the *Enable Application Level Authorization* check box.

**Step 5** Complete the additional fields in the *SIP Trunk Security Profile Configuration* window. For more information on the fields and their configuration options, see Online Help.

**Step 6** Click *Save.*

**Note** You must assign the profile to a trunk in the *Trunk Configuration* window so that the trunk can uses the settings.

Configure Common Device Configuration

A common device configuration comprises a set of optional set of user-specific feature attributes. If you are deploying IPv6, you can use this configuration to assign IPv6 preferences for SIP trunks or SCCP phones.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose *Device > Device Settings > Common Device Configuration.*

**Step 2** Click *Add New.*

**Step 3** For SIP trunks, SIP Phones or SCCP phones, choose a value for the *IP Addressing Mode* drop-down list:
Configure SIP Trunks

Use this procedure to configure a SIP trunk. You can assign up to 16 destination addresses for a SIP trunk.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Trunk.
Step 2 Click Add New.
Step 3 From the Trunk Type drop-down list, choose SIP Trunk.
Step 4 From the Protocol Type drop-down list, choose the type of SIP trunk that matches your deployment and click Next:
   - None (Default)
   - Call Control Discovery
   - Extension Mobility Cross Cluster
   - Cisco Intercompany Media Engine
   - IP Multimedia System Service Control
Step 5 (Optional) If you want to apply a Common Device Configuration to this trunk, select the configuration from the drop-down list.
Step 6 Check the SRTP Allowed check box if you want to allow encrypted media over the trunk.
Step 7 Check the Run on All Active Unified CM Nodes check box if you want to enable the trunk for all cluster nodes.
Step 8 Configure the destination address for the SIP trunk:
   a) In the Destination Address text box, enter an IPv4 address, fully qualified domain name, or DNS SRV record for the server or endpoint that you want to connect to the trunk.
b) If the trunk is a dual stack trunk, in the **Destination Address IPv6** text box, enter an IPv6 address, fully qualified domain name, or DNS SRV record for the server or endpoint that you want to connect to the trunk.

c) If the destination is a DNS SRV record, check the **Destination Address is an SRV** check box.

d) To add additional destinations, click the (+).

**Step 9**  
From the **SIP Trunk Security Profile** drop-down, assign a security profile. If you don't select this option, a nonsecure profile will be assigned.

**Step 10**  
From the **SIP Profile** drop-down list, assign a SIP profile.

**Step 11**  
(Optional) If you want to assign a normalization script to this SIP trunk, from the **Normalization Script** drop-down list, select the script that you want to assign.

**Step 12**  
Configure any additional fields in the **Trunk Configuration** window. For more information on the fields and their configuration options, see Online Help.

**Step 13**  
Click **Save**.
CHAPTER 15

Configure H.323 Trunks

- H.323 Trunk Overview, on page 101
- H.323 Trunk Prerequisites, on page 102
- Configure H.323 Trunks, on page 102

H.323 Trunk Overview

If you have an H.323 deployment, H.323 trunks provide connectivity to remote clusters and other H.323 devices, such as gateways. H.323 trunks support most of the audio and video codecs that Unified Communications Manager supports for intra-cluster communications, with the exception of wideband audio and wideband video. H.323 trunks use the H.225 protocol for call control signaling and the H.245 protocol for media signaling.

Within Cisco Unified CM Administration, H.323 trunks can be configured using the Inter-cluster Trunk (Non-Gatekeeper Controlled) trunk type and protocol options.

If you have a non-gatekeeper H.323 deployment, you must configure a separate intercluster trunk for each device pool in the remote cluster that the local Unified Communications Manager can call over the IP WAN. The intercluster trunks statically specify either the IPv4 addresses or hostnames of the remote devices.

You can configure up to 16 destination addresses for a single trunk.

Intercluster Trunks

When configuring intercluster trunk connections between two remote clusters, you must configure an intercluster trunk on each cluster and match the trunk configurations so that the destination addresses used by one trunk match the call processing nodes that are used by the trunk from the remote cluster. For example:

- Remote cluster trunk uses Run on all Active Nodes—The remote cluster trunk uses all nodes for call processing and load balancing. In the local intercluster trunk that originates in the local cluster, add in the IP addresses or hostnames for each server in the remote cluster.

- Remote cluster does not use Run on all Active Nodes—The remote cluster trunk uses the servers from the Unified Communications Manager Group that is assigned to the trunk's device pool for call processing and load balancing. In the local intercluster trunk configuration, you must add the IP address or hostname of each node from the Unified Communications Manager group used by the remote cluster trunk's device pool.
Secure Trunks

To configure secure signaling for H.323 trunks, you must configure IPSec on the trunk. For details, see the Security Guide for Cisco Unified Communications Manager. To configure the trunk to allow media encryption, check the SRTP allowed check box in the Trunk Configuration window.

Note

Gatekeepers are no longer widely used, but you can also configure your H.323 deployment to use gatekeeper-controlled trunks. For details on how to set up gatekeeper-controlled trunks, refer to Cisco Unified Communications Manager Administration Guide, Release 10.0(1).

H.323 Trunk Prerequisites

Plan out your H.323 deployment topology. For intercluster trunks, make sure you know which servers the corresponding remote cluster trunks use for call processing and load balancing. You will have to configure your local intercluster trunk to connect to each call processing server used by the trunk in the remote cluster.

If you are using Cisco Unified Communications Manager groups assigned to a trunk device pool for load balancing on the trunk, complete the configuration in chapter "Configure Trunks", Core Settings for Device Pools Configuration Task Flow section.

Configure H.323 Trunks

Use this procedure to configure trunks for an H.323 deployment.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose Device &gt; Trunk.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Trunk Type drop-down list box, choose Inter-Cluster Trunk (Non-Gatekeeper Controlled).</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Protocol drop-down list box, choose Inter-Cluster Trunk.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the Device Name text box, enter the unique identifier for the trunk.</td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Device Pool drop-down list box, select the device pool that you configured for this trunk.</td>
</tr>
<tr>
<td>Step 7</td>
<td>If you want to use every node in the local cluster for processing for this trunk, check the Run on all Active Unified CM Nodes check box.</td>
</tr>
<tr>
<td>Step 8</td>
<td>If you want to allow encrypted media across the trunk, check the SRTP Allowed check box.</td>
</tr>
<tr>
<td>Step 9</td>
<td>If you want to configure H.235 pass through, check the H.235 Pass Through Allowed check box.</td>
</tr>
<tr>
<td>Step 10</td>
<td>In the Remote Cisco Unified Communications Manager Information section, enter an IP address or hostname for each remote server to which this trunk connects.</td>
</tr>
</tbody>
</table>
CHAPTER 16

Configure SRST

- Survivable Remote Site Telephony Overview, on page 103
- Survivable Remote Site Telephony Configuration Task Flow, on page 104
- SRST Restrictions, on page 107

Survivable Remote Site Telephony Overview

Survivable Remote Site Telephony (SRST) is an optional feature for sites that depend on a Wide Area Network (WAN) connection to a Unified Communications Manager node. SRST references, which are configured in the Unified Communications Manager Administration interface, allow IP gateways to provide limited telephony service to IP phones at the remote site in the event of a WAN outage:

- IP phones at the remote site can call each other
- Calls from the PSTN can reach the IP phones
- Calls from the IP phones can reach the external world through the PSTN

When phones at the remote site lose connectivity to all associated Unified Communications Manager nodes, the phones connect to the SRST reference IP gateway. The status line indication on the IP phone shows the phone has failed over to the backup SRST gateway. When the connection to Unified Communications Manager is restored, the IP phones reregister with Unified Communications Manager and full telephony services are restored.

SRST supports remote sites that may have a mix of SCCP and SIP endpoints in addition to PSTN gateway access.

Connection Monitor Duration

An IP phone that connects to an SRST gateway over a Wide Area Network (WAN) reconnects itself to Unified Communications Manager as soon as it can establish a connection with Unified Communications Manager over the WAN link. However, if the WAN link is unstable, the IP phone switches back and forth between the SRST gateway and Unified Communications Manager. This situation causes temporary loss of phone service (no dial tone). These reconnect attempts, known as WAN link flapping issues, continue until the IP phone successfully reconnects itself to Unified Communications Manager.

To resolve the WAN link flapping issues between Unified Communications Manager and an SRST gateway, you can define the number of seconds (Connection Monitor Duration) that the IP Phone monitors its connection to Unified Communications Manager before it unregisters from the SRST gateway and reregisters to Unified
Communications Manager. The IP phone receives the connection monitor duration value in the XML configuration file.

**Survivable Remote Site Telephony Configuration Task Flow**

**Before you begin**

Examine the dial plan. If there are 7 or 8 digits in the dial plan, you may need to configure translation rules. For more information about translation rules, see Configure Translation Patterns, on page 158.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure an SRST Reference, on page 104</td>
<td>Configure the gateway that can provide limited call control functionality when all other Unified Communications Manager nodes are unreachable.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign the SRST Reference to a Device Pool, on page 105</td>
<td>For each device pool, assign the gateways that calling devices search when they attempt to complete a call if Unified Communications Manager is unavailable.</td>
</tr>
</tbody>
</table>
| Step 3 | Perform one of the following tasks:  
  • Configure Connection Monitor Duration for the Cluster, on page 105  
  • Configure Connection Monitor Duration for a Device Pool, on page 106 | **Optional**: Configure the connection monitor duration. You can apply a cluster-wide default value, or apply the configuration to the devices in a device pool. |
| Step 4 | Enable SRST on the SRST Gateway, on page 106 | Configure SRST parameters on the gateway. |

**Configure an SRST Reference**

An SRST reference comprises the gateway that can provide limited Cisco Unified Communications Manager functionality when all other Cisco Unified Communications Manager nodes for a device are unreachable.

**Procedure**

**Step 1** Log into Cisco Unified CM Administration and choose System > SRST.

**Step 2** Click Add New.

**Step 3** Configure the fields in the **SRST Reference Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 4** Click Save.
Assign the SRST Reference to a Device Pool

You can configure SRST for each device pool of phones. When you assign an SRST reference to a device pool, all phones in the device pool try to connect to the assigned SRST gateway if they cannot reach any Cisco Unified Communications Manager node.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose System > Device Pool.

**Step 2** Click Find and choose the device pool to which the remote IP phones are registered.

**Step 3** In the Roaming Sensitive Settings area, choose the SRST reference from the SRST Reference drop-down list.

The SRST Reference drop-down list contains the following options:

- **Disable**—If a phone cannot reach any Cisco Unified Communications Manager node, it does not try to connect to an SRST gateway.

- **Use Default Gateway**—If a phone cannot reach any Cisco Unified Communications Manager node, it tries to connect to its IP gateway as an SRST gateway.

- **User-Defined**—If a phone cannot reach any Cisco Unified Communications Manager node, it tries to connect to this SRST gateway.

**Step 4** Click Save.

Configure Connection Monitor Duration for the Cluster

This procedure is optional. Complete this procedure only if you want to change the system value (enterprise parameter) for the connection monitor duration.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose System > Enterprise Parameters.

**Step 2** Enter a value in the Connection Monitor Duration field. The default value is 120 seconds. The maximum number of seconds that you can enter in the field is 2592000.

**Step 3** Click Save.

**Note** You must restart all services for the change to take effect.

The enterprise parameter forms the cluster default for the Connection Monitor Duration. However, if an overriding configuration exists within a device pool, that setting overrides the enterprise parameter setting for the devices that use the device pool.
Configure Connection Monitor Duration for a Device Pool

This procedure is optional. Complete this procedure only if the following is true:

- You do not want to use the cluster-wide value for the connection monitor duration.
- You want to define a separate connection monitor duration value for this device pool.

Tip

When you change the value of the connection monitor duration for a device pool, it applies only to the device pool that is being updated. All other device pools use the value in their own Connection Monitor Duration fields or use the cluster-wide value that is configured in the Connection Monitor Duration enterprise parameter.

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Device Pool.
Step 2  Click Find and choose the device pool to which the remote IP phones are registered.
Step 3  In the Roaming Sensitive Settings area, enter a value in the Connection Monitor Duration field. The maximum number of seconds that you can enter in the field is 2592000.

Note  This setting overrides the enterprise parameter setting for connection monitor duration.

Step 4  Click Save.

Enable SRST on the SRST Gateway

Before you begin

- Assign the SRST Reference to a Device Pool, on page 105
- (Optional) Perform one of the following tasks:
  - Configure Connection Monitor Duration for the Cluster, on page 105
  - Configure Connection Monitor Duration for a Device Pool, on page 106

Procedure

Step 1  Log into the SRST gateway (router).
Step 2  Enter the command call-manager-fallback
This command enables SRST on the router.
Step 3  Enter the command max-ephones max-phones, where max-phones is the maximum number of supported Cisco IP phones.
Step 4  Enter the command max-dn max-directory-numbers where max-directory-numbers is the maximum number of directory numbers (DN) or virtual voice ports that can be supported by a router.
Step 5 Enter the command `ip source-address ip-address` where `ip-address` is a preexisting router IP address, typically one of the addresses of the Ethernet port of the router. This command enables the SRST router to receive messages from Cisco IP Phones through the specified IP address.

---

## SRST Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| Deleting SRST References | You cannot delete SRST references that device pools or other items are using. To find out which device pools are using the SRST reference, click the [Dependency Records](#) link from the **SRST Reference Configuration** window. If the dependency records are not enabled for the system, the dependency records summary window displays a message. If you try to delete an SRST reference that is in use, Unified Communications Manager displays an error message. Before you delete an SRST reference that is currently in use, perform either or both of the following tasks:   
  • Assign a different SRST reference to any device pools that are using the SRST reference that you want to delete.   
  • Delete the device pools that are using the SRST reference that you want to delete. |
| **Note**                 | Before you delete an SRST reference, check carefully to ensure that you are deleting the correct SRST reference. You cannot retrieve deleted SRST references. If an SRST reference is accidentally deleted, you must rebuild it. |
PART III

Configure the Dial Plan

• Dial Plan Overview, on page 111
• Configure Partitions, on page 115
• Install a National Numbering Plan, on page 121
• Configure Call Routing, on page 125
• Configure Hunt Pilots, on page 149
• Configure Translation Patterns, on page 157
• Configure Transformation Patterns, on page 159
• Configure Dial Rules, on page 163
• Configure Intercluster Lookup Service, on page 171
• Configure Global Dial Plan Replication, on page 183
• Configure URI Dialing, on page 193
Dial Plan Overview

- About the Dial Plan, on page 111
- Dial Plan Prerequisites, on page 111
- Dial Plan Configuration, on page 111

About the Dial Plan

The dial plan is responsible for instructing the Cisco Unified Communications Manager system about how to route calls. When you configure a dial plan, you define such rules as:

- the type of calls that are allowed
- the preferred path that the system uses to place a call, as well as alternate paths
- how extensions are dialed
- how called and calling numbers are presented

Dial Plan Prerequisites

Before you configure the dial plan, complete these tasks:

- Initial Configuration Task Flow, on page 9
- Inbound and Outbound Calling Configuration, on page 61

Dial Plan Configuration

Complete the following task flows to configure the dial plan for your system.

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Partition Configuration Task Flow, on page 116</td>
<td>Configure partitions to create a logical grouping of directory numbers (DNs) and route</td>
</tr>
</tbody>
</table>
### Configure the Dial Plan

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>patterns with similar reachability characteristics. Partitions facilitate call routing by dividing the route plan into logical subsets that are based on organization, location, and call type.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>National Numbering Plan Installation Task Flow, on page 122</td>
</tr>
<tr>
<td></td>
<td>Optional. Cisco Unified Communications Manager provides a default North American National Numbering Plan (NANP). For countries with different dial plan requirements, you can install a Cisco International Dial Plan and use it to create a unique numbering plan that is specific to your requirements. When you use a national numbering plan, you can configure route patterns that use the @ symbol, along with route filters, to create patterns for local, national, long distance, and emergency calling. Using a national numbering dial plan is optional. If you do not use a national numbering plan, you can configure one manually.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Call Routing Configuration Task Flow, on page 126</td>
</tr>
<tr>
<td></td>
<td>Configure a route plan to route internal calls and external calls to a private network or the public switched telephone network (PSTN).</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Hunt Pilot Configuration Task Flow, on page 149</td>
</tr>
<tr>
<td></td>
<td>Configure a hunt pilot when you want to extend a call to one or more lists of numbers, where each list specifies a hunting order. When a call extends to a hunt party from these lists and the party fails to answer or is busy, hunting resumes with the next hunt party.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Translation Pattern Configuration Task Flow, on page 158</td>
</tr>
<tr>
<td></td>
<td>Configure translation patterns to manipulate inbound numbers from your voice gateway to the Cisco Unified Communications Manager. You can use translation patterns to change the calling and called number before the system forwards the call to the receiving endpoint. This translation is transparent and allows you to map extensions from the public to the private network.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Transformation Pattern Configuration Task Flow, on page 159</td>
</tr>
<tr>
<td></td>
<td>Configure transformation patterns for phones when you want to modify the calling number display on an inbound call. Configure transformation patterns for gateways or trunks when you want to modify the outgoing calling or called number display sent out for outbound calls. You can also use transformation patterns</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>---------------------------------------------------------------------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td></td>
<td>to modify the outbound redirecting number (known as a diversion header in SIP devices).</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td><strong>You can configure different types of dial rules: application dial rules, directory lookup dial rules, and SIP dial rules.</strong></td>
</tr>
<tr>
<td>Dial Rules Configuration Task Flow, on page 164</td>
<td>• Configure application dial rules to add and sort the priority of dialing rules for applications such as Cisco Web Dialer and Cisco Unified Communications Manager Assistant.</td>
</tr>
<tr>
<td></td>
<td>• Configure directory lookup dial rules to transform caller identification numbers into numbers that can be looked up in the directory.</td>
</tr>
<tr>
<td></td>
<td>• Configure the SIP Dial Rules to create dial patterns for phones that are running SIP. This procedure is typically for legacy SIP phones.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td><strong>Configure Intercluster Lookup Service (ILS) to create networks of remote Cisco Unified Communications Manager clusters. You can configure ILS on a pair of clusters and then join those clusters to form an ILS network.</strong></td>
</tr>
<tr>
<td>ILS Configuration Task Flow, on page 172</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td><strong>If you have configured an Intercluster Lookup Service (ILS) network, you can configure global dial plan replication to create a global dial plan that spans across the ILS network and includes intercluster dialing of directory URIs and alternate numbers.</strong></td>
</tr>
<tr>
<td>Global Dial Plan Replication Task Flow, on page 186</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td><strong>Configure URI dialing when you want to route calls to an endpoint using the directory URI as the call address. The directory URI follows the username@host format, where the host portion is an IPv4 address or a fully qualified domain name.</strong></td>
</tr>
<tr>
<td>URI Dialing Configuration Task Flow, on page 195</td>
<td></td>
</tr>
</tbody>
</table>
Configure Partitions

- Partitions Overview, on page 115
- Partition Configuration Task Flow, on page 116
- Partition Interactions and Restrictions, on page 119

Partitions Overview

Partitions are logical groups of any of the following:

- Route patterns
- Directory numbers (DNs)
- Translation patterns
- Transformation patterns
- Universal resource indicators (URIs)
- Hunt pilots

Partitions facilitate call routing by dividing the route plan into logical subsets that are based on similar accessibility requirements, organization, location, and call type.

Class of Service

You can use partitions and calling search spaces (CSS) to configure classes of service. The table below provides an example of partitions and calling search spaces that you can create for classes of service that provide PSTN access to:

- Emergency calls
- Local calls
- National calls
- International dialing
Table 9: Examples of Partitions and Calling Search Spaces

<table>
<thead>
<tr>
<th>Calling Search Space</th>
<th>Route Partition 1</th>
<th>Route Partition 2</th>
<th>Route Partition 3</th>
<th>Capabilities</th>
</tr>
</thead>
<tbody>
<tr>
<td>Base_CSS</td>
<td>Base_PT</td>
<td>—</td>
<td>—</td>
<td>• Emergency</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• On-net</td>
</tr>
<tr>
<td>LocalPSTN_CSS</td>
<td>PSTN_Local_PT</td>
<td>—</td>
<td>—</td>
<td>• Emergency</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• On-net</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• Local</td>
</tr>
<tr>
<td>NationalPSTN_CSS</td>
<td>PSTN_Local_PT</td>
<td>PSTN_National_PT</td>
<td>—</td>
<td>• Emergency</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• On-net</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• Local</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• National</td>
</tr>
<tr>
<td>InternationalPSTN_CSS</td>
<td>PSTN_Local_PT</td>
<td>PSTN_National_PT</td>
<td>PSTN_Intl_PT</td>
<td>• Emergency</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• On-net</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• Local</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• National</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>• International</td>
</tr>
</tbody>
</table>

Devices automatically register with a calling search space such as Base_CSS. This allows all devices to dial both on-net and emergency off-net numbers. You must assign the remaining calling search spaces to the directory number on the user device profile to provide local 7-digit or local 10-digit, national, and international dialing capabilities.

**Partition Configuration Task Flow**

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Partitions, on page 117</td>
<td>Configure partitions to create a logical groupings of system resources with similar reachability characteristics.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Calling Search Spaces, on page 118</td>
<td>Configure the partitions that calling devices search when they are attempting to complete a call.</td>
</tr>
</tbody>
</table>
Configure Partitions

Configure partitions to create a logical group of system resources with similar reachability characteristics. You can create partitions for any of the following:

- Route patterns
- Directory numbers (DNs)
- Translation patterns
- Transformation patterns
- Universal resource indicators (URIs)
- Hunt pilots

Partitions facilitate call routing by dividing the route plan into logical subsets that are based on organization, location, and call type. You can configure multiple partitions.

Procedure

Step 1  From Cisco Unified CM Administration, choose Call Routing > Class of Control > Partition.
Step 2  Click Add New to create a new partition.
Step 3  In the Partition Name, Description field, enter a name for the partition that is unique to the route plan. Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.
Step 4  Enter a comma (,) after the partition name and enter a description of the partition on the same line. The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (< >), or square brackets ([ ]). If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.
Step 5  To create multiple partitions, use one line for each partition entry.
Step 6  From the Time Schedule drop-down list, choose a time schedule to associate with this partition. The time schedule specifies when the partition is available to receive incoming calls. If you choose None, the partition remains active at all times.
Step 7  Select one of the following radio buttons to configure the Time Zone:

  - Originating Device—When you select this radio button, the system compares the time zone of the calling device to the Time Schedule to determine whether the partition is available to receive an incoming call.

  - Specific Time Zone—After you select this radio button, choose a time zone from the drop-down list. The system compares the chosen time zone to the Time Schedule to determine whether the partition is available to receive an incoming call.

Step 8  Click Save.
Partition Name Guidelines

The list of partitions in a calling search space is limited to a maximum of 1024 characters. This means that the maximum number of partitions in a CSS varies depending on the length of the partition names. Use the following table to determine the maximum number of partitions that you can add to a calling search space if partition names are of fixed length.

Table 10: Partition Name Guidelines

<table>
<thead>
<tr>
<th>Partition Name Length</th>
<th>Maximum Number of Partitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>2 characters</td>
<td>340</td>
</tr>
<tr>
<td>3 characters</td>
<td>256</td>
</tr>
<tr>
<td>4 characters</td>
<td>204</td>
</tr>
<tr>
<td>5 characters</td>
<td>172</td>
</tr>
<tr>
<td>...</td>
<td>...</td>
</tr>
<tr>
<td>10 characters</td>
<td>92</td>
</tr>
<tr>
<td>15 characters</td>
<td>64</td>
</tr>
</tbody>
</table>

Configure Calling Search Spaces

A calling search space is an ordered list of route partitions that are typically assigned to devices. Calling search spaces determine the partitions that calling devices can search when they are attempting to complete a call.

Procedure

**Step 1** From Cisco Unified CM Administration, choose Call Routing > Class of Control > Calling Search Space.

**Step 2** Click Add New.

**Step 3** In the Name field, enter a name. Ensure that each calling search space name is unique to the system. The name can include up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).

**Step 4** In the Description field, enter a description. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

**Step 5** From the Available Partitions drop-down list, perform one of the following steps:

- For a single partition, select that partition.
- For multiple partitions, hold down the Control (CTRL) key, then select the appropriate partitions.

**Step 6** Select the down arrow between the boxes to move the partitions to the Selected Partitions field.

**Step 7** (Optional) Change the priority of selected partitions by using the arrow keys to the right of the Selected Partitions box.
Partition Interactions and Restrictions

**Table 11: Partition Restrictions**

<table>
<thead>
<tr>
<th>Function or Action</th>
<th>Restriction</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Delete a Partition</td>
<td>Ensure that you complete one of the following tasks, before you delete a partition:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Assign a different partition to any calling search spaces, devices, or other items that are using the partition that you want to delete.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Delete the calling search spaces, devices, or other items that are using the partition that you want to delete.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Check carefully to ensure that you are deleting the correct partition, because you cannot retrieve deleted partitions. If you accidentally delete a partition, you must rebuild it.</td>
<td></td>
</tr>
<tr>
<td>Time of Day Routing</td>
<td>Configure a schedule for when a partition is available to accept incoming calls. For details on configuring time of day routing, see the Configure Call Routing chapter.</td>
<td></td>
</tr>
<tr>
<td>Logical Partitioning</td>
<td><strong>Optional</strong>: Allows you to split your internal VoIP network from your external network with gateway and trunk access. Logical partitioning is optional for most deployments, but is mandatory in countries such as India where regulations mandate that all calls that leave the internal network go to a local PSTN gateway. For details on Configuring Logical Partitioning, refer to the &quot;Configure Logical Partitioning&quot; section in the Feature Configuration Guide for Cisco Unified Communication Manager.</td>
<td></td>
</tr>
</tbody>
</table>

**Partition Restrictions**

**Table 12: Partition Restrictions**

<table>
<thead>
<tr>
<th>Function or Action</th>
<th>Restriction</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Delete a Partition</td>
<td>Ensure that you complete one of the following tasks, before you delete a partition:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Assign a different partition to any calling search spaces, devices, or other items that are using the partition that you want to delete.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Delete the calling search spaces, devices, or other items that are using the partition that you want to delete.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Check carefully to ensure that you are deleting the correct partition, because you cannot retrieve deleted partitions. If you accidentally delete a partition, you must rebuild it.</td>
<td></td>
</tr>
</tbody>
</table>
### Function or Action | Restriction
--- | ---
Translation Patterns | A translation pattern contains digit manipulations and is assigned to a partition. When a call matches the translation pattern, Unified CM performs the translation and then reroutes the call using the calling search space that the translation pattern specifies. For details on translation patterns, see the Configure Call Routing chapter.

Time of Day Routing | Configures a schedule for when a partition is available to accept incoming calls. For details on configuring time of day routing, see the Configure Call Routing chapter.

Logical Partitioning | **Optional**: Allows you to split your internal VoIP network from your external network with gateway and trunk access. Logical partitioning is optional for most deployments, but is mandatory in countries such as India where regulations mandate that all calls that leave the internal network go to a local PSTN gateway. For details on configuring logical partitioning, refer to the Logical Partitioning chapter in the Feature Configuration Guide.
CHAPTER 19

Install a National Numbering Plan

• National Numbering Plan Overview, on page 121
• National Numbering Plan Prerequisites, on page 121
• National Numbering Plan Installation Task Flow, on page 122

National Numbering Plan Overview

Unified Communications Manager provides a default North American Numbering Plan (NANP). For countries with different dial plan requirements, you can install a Cisco International Dial Plan and use it to create a unique numbering plan that is specific to your requirements.

The numbering plan contains the Discard Digits Instructions (DDIs) and tags specific to that numbering plan. You can use these items when configuring call routing to create routing rules that are applicable to the numbering plan.


National Numbering Plan Prerequisites

If you are installing a National Numbering Plan for countries outside of North America, download the Cisco Option Package (COP) file that contains the international dial plans for the current release. The COP file uses the naming convention IDP v.x, and is available from the Cisco website:

• https://software.cisco.com/download/navigator.html

Place the file on an external FTP or SFTP server that Unified Communications Manager can access.
# National Numbering Plan Installation Task Flow

## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Install the COP file, on page 122</td>
<td>Optional. To install a numbering plan for countries outside of North America, download the Cisco Option Package (COP) file that contains the international dial plans for the current release.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Install a National Numbering Plan, on page 123</td>
<td>Install the national numbering plan on each Unified Communications Manager node in the cluster. Perform this procedure only if you are installing a National Numbering Plan for countries outside of North America.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Restart the CallManager Service, on page 123</td>
<td>The changes take effect after you restart the service.</td>
</tr>
</tbody>
</table>

## Install the COP file

Use this procedure to install a Cisco Option Package (COP) file that contains international dial plans.

### Procedure

1. **Step 1** Begin this procedure on the Unified Communications Manager publisher node. From Cisco Unified Communications OS Administration, choose Software Upgrades > Install. The Software Installation/Upgrade window appears.
2. **Step 2** In the Source field, choose Remote File System.
3. **Step 3** Configure the fields on the Software Installation/Upgrade window. See the Related Topics for more information about the fields and their configuration options.
4. **Step 4** Click Next. The window refreshes with a list of available software options and upgrades.
5. **Step 5** From the Options/Upgrades drop-down list, choose the DP COP file and click Next. The Installation File window opens and downloads the file from the FTP server. The window displays the progress of the download.
6. **Step 6** When the Checksum window appears, verify the checksum value against the checksum for the file that you downloaded.
7. **Step 7** Click Next to proceed with the software upgrade. A warning message displays the DP COP file that you selected to install.
8. **Step 8** Click Install. The Install Status window appears.
9. **Step 9** Click Finish.
Step 10  Repeat this procedure on the Unified Communications Manager subscriber nodes. You must install the COP file on all the nodes in the cluster.

Related Topics
COP File Installation Fields, on page 123

COP File Installation Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory</td>
<td>Enter the directory where the COP file is located.</td>
</tr>
<tr>
<td>Remote Server</td>
<td>Enter the host name or IP address of the server where COP file is located.</td>
</tr>
<tr>
<td>Remote User</td>
<td>Enter the user name for the remote server.</td>
</tr>
<tr>
<td>Remote Password</td>
<td>Enter the password for the remote server.</td>
</tr>
<tr>
<td>Transfer Protocol</td>
<td>Select a protocol to use when connecting with the remote server.</td>
</tr>
</tbody>
</table>

Install a National Numbering Plan

Perform this procedure only if you are installing a national numbering plan for countries outside of North America.

Install the national numbering plan on each Unified Communications Manager node in the cluster. Begin with the Unified Communications Manager publisher node.

Procedure

Step 1  From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Plan Installer.
Step 2  Enter search criteria and click Find.
Step 3  Choose the dial plan version that you want to install from the Available Version drop-down list.
Step 4  Click Install. The Status displays that the dial plan has been installed.
Step 5  Repeat this procedure for every subscriber node in the cluster.

Restart the CallManager Service

Procedure

Step 1  From the Cisco Unified Serviceability interface, choose Tools > Control Center - Feature Services.
Step 2  Choose the Unified Communications Manager server from the Servers drop-down list. In the CM Services area, Cisco CallManager displays in the Service Name column.
Step 3  Click the radio button that corresponds to the Cisco CallManager service.
Step 4  Click Restart.
The service restarts and displays the message, Service Successfully Restarted.
Configure Call Routing

Call Routing Overview

The system uses route plans to determine how to route calls between clusters, and how to route external calls to a private network or to the Public Switched Telephone Network (PSTN). The route plan that you configure specifies the path that the system uses to route each type of call. For example, you can create a route plan that uses the IP network for on-net calls, or that uses one carrier for local PSTN calls and another for international calls.

The system has a three-tiered approach to route planning that uses the following components:

- **Route Patterns**—The system searches for a configured route pattern that matches the external dialed string and uses it to select a gateway or a corresponding route list.
- **Route Lists**—A prioritized list of the available paths for the call.
- **Route Groups**—The available paths; the route group distributes the call to gateways and trunks.

In addition to these building blocks, the route plan can also include the following components:

- **Local Route Groups**—Decouple the location of a PSTN gateway from the route patterns that are used to access the gateway.
- **Route Filters**—Restrict certain numbers that are otherwise allowed by the route pattern.
- **Automated Alternate Routing**—Automatically reroute calls through the PSTN or other network when the system blocks a call due to insufficient bandwidth.
- **Time-of-day Routing**—Create a time schedule that specifies when a partition is available to receive incoming calls.
Call Routing Prerequisites

- Complete the tasks in the Partition Configuration Task Flow, on page 116.
- Ensure that you have the following information:
  - Internal number extensions
  - A plan listing the calls that route to each gateway

Call Routing Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Local Route Groups, on page 131</td>
<td>Optional. Configure local route groups to reduce the number of route lists that you need. Route lists point to the PSTN gateway that the system uses to route the call, based on the location of the PSTN gateway. As an alternative, you can use local route groups to decouple the location of a PSTN gateway from the route patterns that are used to access the gateway. This configuration allows phones and other devices from different locations to use a single set of route patterns, while Unified Communication Manager selects the correct gateway to route the call.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Route Groups, on page 130</td>
<td>Optional. Configure route groups to set the selection order of the gateway devices. Route groups contain one or more devices.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Route Lists, on page 131</td>
<td>Optional. Route lists contain one or more route groups. Configure route lists to control the selection order of the route groups. If you configure a route list, you must configure at least one route group.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Route Filters, on page 133</td>
<td>Optional. Use route filters to restrict certain numbers that are otherwise allowed by a route pattern. Route filters are mandatory if you are using a dial plan installer; that is, if you install a dial plan file and then configure a route pattern based on that numbering plan. Route filters are optional if you are configuring a dial plan manually.</td>
</tr>
</tbody>
</table>
### Configure Route Patterns

Unified Communications Manager uses route patterns to route or block internal and external calls. You can assign route patterns to gateways, to trunks, or to a route list that contains one or more route groups.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>If you are configuring a dial plan manually, you need to configure route filters whenever you have a route pattern that contains the @ wildcard. When the route pattern contains the @ wildcard, the system routes calls according to the numbering plan that you specify with a route filter.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>Configure Route Patterns, on page 127</strong></td>
</tr>
<tr>
<td>Configure route patterns to direct calls to specific devices and to include or exclude specific digit patterns. You can assign route patterns to gateways, to trunks, or to a route list that contains one or more route groups.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>Configure Time of Day Routing, on page 137</strong></td>
</tr>
<tr>
<td>Optional. Create a time schedule that specifies when a partition is available to receive incoming calls.</td>
<td></td>
</tr>
</tbody>
</table>

### Procedure

1. **Step 1** From Cisco Unified CM Administration, choose **Call Routing > Route/Hunt > Route Pattern**.
2. **Step 2** Perform one of the following:
   - Click **Add New** to create a new route pattern.
   - Click **Find** and select an existing route pattern.

   The **Route Pattern Configuration** Window appears.

3. **Step 3** In the **Route Pattern** field, enter the number pattern that the dial string must match.
4. **Step 4** From the **Gateway/Route** drop-down list, select the destination where you want to send calls that match this route pattern.
5. **Step 5** Complete the remaining fields in the **Route Pattern Configuration** window. For more information on the fields and their configuration options, see the system Online Help.
6. **Step 6** Click **Save**.
Wildcards and Special Characters in Route Patterns

Wildcards and special characters in route patterns allow a single route pattern to match a range of numbers (addresses). Use these wildcards and special characters also to build instructions that enable the Unified Communications Manager to manipulate a number before sending it to an adjacent system.

The following table describes the wildcards and special characters that Unified Communications Manager supports.

Table 13: Wildcards and Special Characters

<table>
<thead>
<tr>
<th>Character</th>
<th>Description</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>@</td>
<td>The at symbol (@) wildcard matches all National Numbering Plan numbers. Each route pattern can have only one @ wildcard.</td>
<td>The route pattern 9.@ routes or blocks all numbers that the National Numbering Plan recognizes. The following route patterns examples show National Numbering Plan numbers that the @ wildcard encompasses: • 0 • 1411 • 19725551234 • 101028819725551234 • 01133123456789</td>
</tr>
<tr>
<td>X</td>
<td>The X wildcard matches any single digit in the range 0 through 9.</td>
<td>The route pattern 9XXX routes or blocks all numbers in the range 9000 through 9999.</td>
</tr>
<tr>
<td>!</td>
<td>The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.</td>
<td>The route pattern 91! routes or blocks all numbers in the range 910 through 9199999999999999999999999</td>
</tr>
<tr>
<td>?</td>
<td>The question mark (?) wildcard matches zero or more occurrences of the preceding digit or wildcard value.</td>
<td>The route pattern 91X? routes or blocks all numbers in the range 91 through 9199999999999999999999999999999</td>
</tr>
<tr>
<td>+</td>
<td>The plus sign (+) wildcard matches one or more occurrences of the preceding digit or wildcard value.</td>
<td>The route pattern 91X+ routes or blocks all numbers in the range 910 through 919999999999999999999999999999999</td>
</tr>
<tr>
<td>[ ]</td>
<td>The square bracket ([ ]) characters enclose a range of values.</td>
<td>The route pattern 813510[012345] routes or blocks all numbers in the range 813510 through 8135105.</td>
</tr>
<tr>
<td>-</td>
<td>The hyphen (-) character, used with the square brackets, denotes a range of values.</td>
<td>The route pattern 813510[0-5] routes or blocks all numbers in the range 813510 through 8135105.</td>
</tr>
<tr>
<td>Character</td>
<td>Description</td>
<td>Examples</td>
</tr>
<tr>
<td>-----------</td>
<td>-------------</td>
<td>----------</td>
</tr>
<tr>
<td>^</td>
<td>The circumflex (^) character, used with the square brackets, negates a range of values. Ensure that it is the first character following the opening bracket ([). Each route pattern can have only one ^ character.</td>
<td>The route pattern 813510[^0-5] routes or blocks all numbers in the range 8135106 through 8135109.</td>
</tr>
<tr>
<td>.</td>
<td>The dot (.) character, used as a delimiter, separates the Cisco Unified Communications Manager access code from the directory number. Use this special character, with the discard digits instructions, to strip off the Cisco Unified Communications Manager access code before sending the number to an adjacent system. Each route pattern can have only one dot (.) character.</td>
<td>The route pattern 9.@ identifies the initial 9 as the Cisco Unified Communications Manager access code in a National Numbering Plan call.</td>
</tr>
<tr>
<td>*</td>
<td>The asterisk (*) character can provide an extra digit for special dialed numbers.</td>
<td>You can configure the route pattern *411 to provide access to the internal operator for directory assistance.</td>
</tr>
<tr>
<td>#</td>
<td>The octothorpe (#) character generally identifies the end of the dialing sequence. Ensure the # character is the last character in the pattern.</td>
<td>The route pattern 90118191055# routes or blocks an international number that is dialed from within the National Numbering Plan. The # character after the last 5 identifies this digit as the last digit in the sequence.</td>
</tr>
<tr>
<td>\+</td>
<td>A plus sign preceded by a backslash, that is, \+, indicates that you want to configure the international escape character +. Using \+ means that the international escape character + is used as a dialable digit, not as a wildcard.</td>
<td></td>
</tr>
</tbody>
</table>

**Example of Pre-dot Digit Removal**

One example of using pre-dot digit removal in a route pattern is when you want the phone users to dial an access code to reach an outside line. In North America, users typically dial 9 to access an outside line. You can specify using the following route patterns:

- Local calls: 9 . @ or 9 . [2-9] XXXXX
- International calls: 9 . 0111 #
In these patterns, 9 is the access code for an external line, and the dot (.) is a separator that helps format the route pattern by indicating which digits are internal to the network, and which ones are outside digits. When the system sends the dialed digits to the PSTN, you can use the Discard Digits option to strip the pre-dot digit from the dialed string so that the PSTN can route the call.

**Example of Digit Prefixing**

One example of using digit prefixing in a route pattern is when you configure On-Net dialing between sites. You can create a route pattern so that users within your organization dial 8 + XXX-XXXX to call between sites. For Off-Net calls, you can remove the prefix digit (8) and add a new prefix of 1<area code> so that you can route the call to the PSTN in E.164 format.

**Example of On-Net and Off-Net Patterns**

You can configure a route pattern as OnNet or OffNet using the Call Classification field. You can classify calls as Off-Net in cases where you want your users to get a secondary dial tone to let them know that their call is going outside your organization. For example, if you create a route pattern that requires users to dial 9 to access an outside line, and you classify it as an Off-Net pattern, the system provides the following dial tones:

- A dial tone when the phone is off-hook, before you dial 9.
- A secondary dial tone, after you dial 9 to indicate that the system is ready to call the Public Switched Telephone Network (PSTN) number.

Ensure that you deselect the Allow Device Override check box when you use this option.

**Example of Block and Route Patterns**

Use block and route patterns to prevent outgoing or incoming calls that you do not want to route. Use block patterns to:

- Block specific patterns. For example, blocking the pattern 91900XXXXXXXX prevents users from placing calls to 900 services.
- Prevent toll fraud by blocking calls to specific area codes and locations.

**Configure Route Groups**

Configure a route group to prioritize the order in which the system selects gateways for outgoing calls. Use this procedure to group together gateways that have similar characteristics, so that any gateway in the group can dial the call. The system selects the gateway to use based on the order that you specify when you configure the route group.

You can assign a device to multiple route groups.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Route Group.

The Route Group Configuration window appears.
Configure the Dial Plan

Configure Route Lists

Configure a route list to identify a set of route groups and place them in priority order. Unified Communications Manager uses the order in the route list to search for available devices for outgoing calls.

If you configure a route list, you must configure at least one route group. A route list can contain only route groups and local route groups.

Configure Local Route Groups

Optional. You can configure local route groups to reduce the number of route lists that you need. Route lists point to the PSTN gateway that the system uses to route the call, based on the location of the PSTN gateway. As an alternative, you can use local route groups to decouple the location of a PSTN gateway from the route patterns that are used to access the gateway. This configuration allows phones and other devices from different
locations to use a single set of route patterns, while Cisco Unified Communication Manager selects the correct
gateway to route the call.

For example, a local route group allows you to have a single dial plan for a whole country rather than have
separate dial plans for every city in the country. This approach works for centralized call-deployment scenarios
only.

Note

Extend and Connect works only with Standard Local Route Groups when Local route group for redirected
calls are set to Local route group of last redirecting party.

Mobile Voice Access works only with Standard Local Route Groups when Local route group for redirected
calls are set to Local route group of calling party.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>Configure Local Route Group Names, on page 132</strong></td>
</tr>
</tbody>
</table>

**Step 2** | **Associate a Local Route Group with a Device Pool, on page 133** | To ensure that each device in the system is provisioned to know its local route group, associate the local route group with a device pool. |

**Step 3** | **Add Local Route Group to a Route List, on page 133** | Optional. Configure a local route group that you can add to your route list. When you create a local route group, the system routes outgoing calls to the gateways that are defined for the user at the device pool level. |

**Configure Local Route Group Names**

Optional. The system provides a default local route group called Standard Local Route Group, but you can configure additional local route groups. Use this procedure to name the additional local route groups.

**Procedure**

**Step 1** | From Cisco Unified CM Administration, choose **Call Routing > Route/Hunt > Local Route Group Names**. |
**Step 2** | Click **Add Row**. |
**Step 3** | Enter a name and description for the new local route group. |
**Step 4** | Click **Save**.
Associate a Local Route Group with a Device Pool

You can assign a local route group to use an existing route group, based on the device pool setting of the originating device. This configuration allows phones and other devices from different locations to use a single set of route patterns, while Unified Communication Manager selects the correct gateway to route the call.

To ensure that each device in the system is provisioned to know its local route group, associate the local route group with a device pool.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Device Pool.
Step 2 Enter search criteria, click Find, and select a device pool from the resulting list.
Step 3 In the Local Route Group Settings area, select a route group from the Standard Local Route Group drop-down list.
Step 4 Click Save.

Add Local Route Group to a Route List

Configure a local route group that you can add to your route list. When you create a local route group, the system routes outgoing calls to the gateways that are defined for the user at the device pool level.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Route List.
Step 2 Choose one of the following options:
   • Click Add New button to add a new route list.
   • Click Find and select a route list from the resulting list, to modify the settings for an existing route list.

The Route List Configuration window appears.
Step 3 To add a local route group to the route list, click the Add Route Group button.
Step 4 From the Route Group drop-down list, select a local route group to add to the route list. You can add the standard local route group, or you can add a custom local route group that you have created.
Step 5 Click Save.
Step 6 Click Apply Config.

Configure Route Filters

Route filters use dialed-digit strings to determine how a call is handled. Route filters apply only when you configure a route pattern that contains the @ wildcard. When the route pattern contains the @ wildcard, Unified Communications Manager routes calls according to the numbering plan that you specify in this procedure.
Route filters are mandatory if you are using a dial plan installer; that is, if you install a dial plan file and then configure a route pattern based on that numbering plan. Route plans are optional when configuring dial plans manually.

If you are configuring a dial plan manually, you need to configure route filters whenever you have a route pattern that contains the @ wildcard. When the route pattern contains the @ wildcard, the system routes calls according to the numbering plan that you specify with a route filter.

**Note**
When configuring your call routing, ensure that you do not assign a single route filter to many route patterns. A system core could result if you were to edit a route filter that has hundreds of associated route patterns. This is due to the extra system processing that is required to update call routing for all of the route patterns that use the route filter. Create duplicate route filters and associate any single route filter with no more than 250 Route Patterns.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Call Routing > Route Filter**.

**Step 2**
From the **Numbering Plan** drop-down list, choose a dial plan and click **Next**.

**Step 3**
Enter a name in the **Route Filter Name** field. Ensure each route filter name is unique to the route plan.

**Step 4**
Choose the route filter tags and operators and enter the data to create a clause for this route filter.

For more information about available route filter tags, see **Route Filter Tags**, on page 134.

**Note**
Do not enter route filter tag values for tags that are using the operators EXISTS, DOES-NOT-EXIST, or NOT-SELECTED.

**Step 5**
Choose the route filter operators and enter data, where appropriate, to create a clause for this route filter.

For more information about available route filter operators, see **Route Filter Operators**, on page 136.

**Step 6**
Click **Save**.

**Step 7**
Click **Apply Config**.

**Route Filter Tags**

The tag serves as the core component of a route filter. A tag applies a name to a subset of the dialed-digit string. For example, the NANP number 972-555-1234 comprises LOCAL-AREA-CODE (972), OFFICE-CODE (555), and SUBSCRIBER (1234) route filter tags.

Route filter tags require operators and can require additional values to decide which calls are filtered.

The values for route filter tag fields can contain the wildcard characters X, *, #, [, ], -, ^, and the numbers 0 through 9. The descriptions in the following table use the notations [2-9] and XXXX to represent actual digits. In this notation, [2-9] represents any single digit in the range 2 through 9, and X represents any single digit in the range 0 through 9. Therefore, the three-digit area code in the form [2-9]XX means that you can enter the actual digits 200 through 999, or all wildcards, or any mixture of actual digits and wildcards that results in a pattern with that range.
Route filter tags vary depending on the numbering plan that you choose from the Numbering Plan drop-down list box on the Route Filter Configuration window. The following table describes the route filter tags for the North American Numbering Plan.

**Table 14: Route Filter Tags**

<table>
<thead>
<tr>
<th>Tag</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>AREA-CODE</td>
<td>This three-digit area code in the form [2-9]XX identifies the area code for long-distance calls.</td>
</tr>
<tr>
<td>COUNTRY CODE</td>
<td>These one-, two-, or three-digit codes specify the destination country for international calls.</td>
</tr>
<tr>
<td>END-OF-DIALING</td>
<td>This single character identifies the end of the dialed-digit string. The # character serves as the end-of-dialing signal for international numbers that are dialed within the NANP.</td>
</tr>
<tr>
<td>INTERNATIONAL ACCESS</td>
<td>This two-digit access code specifies international dialing. Calls that originate in the U.S. use 01 for this code.</td>
</tr>
<tr>
<td>INTERNATIONAL DIRECT-DIAL</td>
<td>This one-digit code identifies a direct-dialed international call. Calls that originate in the U.S. use 1 for this code.</td>
</tr>
<tr>
<td>INTERNATIONAL OPERATOR</td>
<td>This one-digit code identifies an operator-assisted international call. This code specifies 0 for calls that originate in the U.S.</td>
</tr>
<tr>
<td>LOCAL-AREA-CODE</td>
<td>This three-digit local area code in the form [2-9]XX identifies the local area code for 10-digit local calls.</td>
</tr>
<tr>
<td>LOCAL-DIRECT-DIAL</td>
<td>This one-digit code identifies a direct-dialed local call. NANP calls use 1 for this code.</td>
</tr>
<tr>
<td>LOCAL-OPERATOR</td>
<td>This one-digit code identifies an operator-assisted local call. NANP calls use 0 for this code.</td>
</tr>
<tr>
<td>LONG-DISTANCE DIRECT-DIAL</td>
<td>This one-digit code identifies a direct-dialed, long-distance call. NANP calls use 1 for this code.</td>
</tr>
<tr>
<td>LONG-DISTANCE OPERATOR</td>
<td>These one- or two-digit codes identify an operator-assisted, long-distance call within the NANP. Operator-assisted calls use 0 for this code, and operator access uses 00.</td>
</tr>
<tr>
<td>NATIONAL-NUMBER</td>
<td>This tag specifies the nation-specific part of the digit string for an international call.</td>
</tr>
<tr>
<td>OFFICE-CODE</td>
<td>This tag designates the first three digits of a seven-digit directory number in the form [2-9]XX.</td>
</tr>
<tr>
<td>SATELLITE-SERVICE</td>
<td>This one-digit code provides access to satellite connections for international calls.</td>
</tr>
<tr>
<td>SERVICE</td>
<td>This three-digit code designates services such as 911 for emergency, 611 for repair, and 411 for information.</td>
</tr>
<tr>
<td>SUBSCRIBER</td>
<td>This tag specifies the last four digits of a seven-digit directory number in the form XXXX.</td>
</tr>
</tbody>
</table>
Route Filter Operators

Route filter tag operators determine whether a call is filtered based on the dialed-digit string that is associated with that tag. The operators EXISTS and DOES-NOT-EXIST simply check for the existence of that part of the dialed-digit string. The operator == matches the actual dialed digits with the specified value or pattern.

The following table describes the operators that you can use with route filter tags.

<table>
<thead>
<tr>
<th>Operator</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>NOT-SELECTED</td>
<td>Specifies do not filter calls based on the dialed-digit string that is associated with this tag.</td>
</tr>
<tr>
<td></td>
<td>Note The presence or absence of the tag with which the operator is associated does not prevent Cisco Unified Communications Manager from routing the call.</td>
</tr>
<tr>
<td>EXISTS</td>
<td>Specifies filter calls when the dialed-digit string that is associated with this tag is found.</td>
</tr>
<tr>
<td></td>
<td>Note Cisco Unified Communications Manager routes or blocks the call only if the dialed-digit string contains a sequence of digits that are associated with the tag.</td>
</tr>
<tr>
<td>DOES-NOT-EXIST</td>
<td>Specifies filter calls when the dialed-digit string that is associated with this tag is not found.</td>
</tr>
<tr>
<td></td>
<td>Note Cisco Unified Communications Manager routes or blocks the call only if the dialed-digit string does not contain a sequence of digits that are associated with the tag.</td>
</tr>
<tr>
<td>==</td>
<td>Specifies filter calls when the dialed-digit string that is associated with this tag matches the specified value.</td>
</tr>
<tr>
<td></td>
<td>Note Cisco Unified Communications Manager routes or blocks the call only if the dialed-digit string contains a sequence of digits that are associated with the tag and within the numbering range that is specified in the attached field.</td>
</tr>
</tbody>
</table>
Route Filter Examples

Example 1: A route filter that uses AREA-CODE and the operator DOES-NOT-EXIST selects all dialed-digit strings that do not include an area code.

Example 2: A route filter that uses AREA-CODE, the operator ==, and the entry 515 selects all dialed-digit strings that include the 515 area code.

Example 3: A route filter that uses AREA-CODE, the operator ==, and the entry 5[2-9]X selects all dialed-digit strings that include area codes in the range of 520 through 599.

Example 4: A route filter that uses TRANSIT-NETWORK, the operator ==, and the entry 0288 selects all dialed-digit strings with the carrier access code 1010288.

Configure Time of Day Routing

Optional. Create a time schedule that specifies when a partition is available to receive incoming calls.

Note
Time of Day routing is not implemented for Message Waiting Indication (MWI) intercept.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure a Time Period, on page 137</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure a Time Schedule, on page 138</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Associate a Time Schedule with a Partition, on page 138</td>
</tr>
</tbody>
</table>

Configure a Time Period

Use this procedure to define time periods. You can define a start time and an end time, and also specify repetition interval either as days of the week or a specified date on the yearly calendar.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Time Period.
Configure a Time Schedule

Use this procedure to create a schedule. The time periods that you configured in the previous procedure are building blocks for this schedule. You can assign time periods to multiple schedules.

**Procedure**

1. From Cisco Unified CM Administration, choose *Call Routing* > *Class of Control* > *Time Schedule*.
2. Configure the fields in the *Time Schedule Configuration* window. For more information on the fields and their configuration options, see the system Online Help.
3. Click **Save**.

Associate a Time Schedule with a Partition

Associate time schedules with partitions to determine where calling devices search when they are attempting to complete a call during a particular time of day.

**Procedure**

1. From Cisco Unified CM Administration, choose *Call Routing* > *Class of Control* > *Partition*.
2. From the *Time Schedule* drop-down list, choose a time schedule to associate with this partition. The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.
3. Click **Save**.

Call Routing Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Filter Associations</td>
<td>When configuring your call routing, be careful not to assign a single route filter to too many route patterns. A system core crash could result if you were to edit a route filter that has hundreds of associated route patterns. This is due to the extra system processing that is required to update call routing for all of the route patterns that use the route filter. Create duplicate route filters to ensure that this does not happen.</td>
</tr>
</tbody>
</table>
Restriction Feature

External Call Control

External call control lets an adjunct route server make call routing decisions for Unified Communications Manager by using the Cisco Unified Routing Rules Interface. When you configure external call control, Unified Communications Manager issues a route request that contains the calling party and called party information to the adjunct route server. That server receives the request, applies appropriate business logic, and returns a route response that instructs your system on how to route the call along with any additional call treatment to apply.

For details, see the Configure External Call Control chapter of the Feature Configuration Guide for Cisco Unified Communications Manager.

Call Control Discovery

With Call Control Discovery, Unified Communications Manager clusters can automatically exchange the DN ranges they host by subscribing to a Cisco IOS service routing protocol called the Service Advertisement Framework (SAF). This feature enables clusters to advertise their own hosted DN ranges into the network as well as to subscribe to advertisements that are generated by other call agents in the network.

The main benefits of using SAF CCD are:

- Automated distribution of call routing information between call agents participating in the same SAF CCD network, thus avoiding incremental configuration work when new call agents are added or when new DN ranges are added to a call agent.
- No reliance on a centralized dial plan resolution control point.
- Automated recovery of inter-call agent call routing information when routing changes occur, including when multiple Unified CM clusters are combined.

To configure Call Control Discovery, refer to the Configure Call Control Discovery chapter of the Feature Configuration Guide for Cisco Unified Communications Manager.

Route Plan Report

You can view a detailed route plan within the Route Plan Report window of Cisco Unified CM Administration (Call Routing > Route Plan Report). The route plan report allows you to view either a partial or full list of your route plan and to go directly to the associated configuration windows by clicking the entry in the Pattern/Directory Number, Partition, or Route Detail columns of the report.

In addition, the route plan report allows you to save report data into a .csv file that you can import into other applications. The .csv file contains more detailed information than the web pages, including directory numbers for phones, route patterns, pattern usage, device name, and device description.

Line Group Setup

This chapter provides information to add or delete a line group or to add directory numbers to or to remove directory numbers from a line group.

For additional information, see topics related to understanding route plans in the Cisco Unified Communications Manager System Guide.
About Line Group Setup

In Cisco Unified Communications Manager Administration, use the Call Routing > Route/Hunt > Line Group menu path to configure line groups.

A line group allows you to designate the order in which directory numbers are chosen. Cisco Unified Communications Manager distributes a call to idle or available members of a line group based on a call distribution algorithm and on the Ring No Answer Reversion (RNAR) Timeout setting.

---

**Note**

Users cannot pick up calls to a DN that belongs to a line group by using the Directed Call Pickup feature.

---

**Tip**

Although you can configure an empty line group with no members (directory numbers), Cisco Unified Communications Manager does not support this configuration for routing calls. If the line group contains no members, the hunt list stops hunting when the call gets routed to the empty line group. To avoid this situation, make sure that you configure at least one member in the line group.

---

**Line Group Configuration Tips**

You must define one or more directory numbers before configuring a line group.

After you configure or update a line group, you can add or remove members from that line group.

---

**Line Group Deletion**

You can delete a line group that one or more route/hunt lists references. If you try to delete a line group that is in use, Cisco Unified Communications Manager displays an error message.

---

**Tip**

Dependency Records is not supported for line groups. As a best practice, always check the configuration before you delete a line group.

---

**Line Group Settings**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line Group Information</td>
<td></td>
</tr>
</tbody>
</table>
Configure the Dial Plan

Line Group Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line Group Name</td>
<td>Enter a name for this line group. The name can comprise up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_). Ensure that each line group name is unique to the route plan. Timesaver: Use concise and descriptive names for your line groups. The Companynamelocationgroup format usually provides a sufficient level of detail and is short enough to enable you to quickly and easily identify a line group. For example, CiscoDallasAA1 identifies a Cisco Access Analog line group for the Cisco office in Dallas.</td>
</tr>
<tr>
<td>RNA Reversion Timeout</td>
<td>Enter a time, in seconds, after which Unified Communications Manager will distribute a call to the next available or idle member of this line group or to the next line group if the call is not answered and if the first hunt option, Try next member; then, try next group in Hunt List, is chosen. The RNA Reversion Timeout applies at the line-group level to all members.</td>
</tr>
</tbody>
</table>
Choose a distribution algorithm, which applies at the line-group level, from the options in the drop-down list box:

- **Top Down**—If you choose this distribution algorithm, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member.

- **Circular**—If you choose this distribution algorithm, Unified Communications Manager distributes a call to idle or available members starting from the \((n+1)\)th member of a route group, where the \(n\)th member is the next sequential member in the list who is either idle or busy but not “down.” If the \(n\)th member is the last member of a route group, Unified Communications Manager distributes a call starting from the top of the route group.

- **Longest Idle Time**—If you choose this distribution algorithm, Unified Communications Manager only distributes a call to idle members, starting from the longest idle member to the least idle member of a line group.

- **Broadcast**—If you choose this distribution algorithm, Unified Communications Manager distributes a call to all idle or available members of a line group simultaneously. See the Note in the description of the Selected DN/Route Partition field for additional limitations in using the Broadcast distribution algorithm.

The default value specifies Longest Idle Time.
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| No Answer | For a given distribution algorithm, choose a hunt option for Unified Communications Manager to use if a call is distributed to a member of a line group that does not answer. This option gets applied at the member level. Choose from the options in the drop-down list box:  
- Try next member; then, try next group in Hunt List—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. If unsuccessful, Unified Communications Manager then tries the next line group in a hunt list.  
- Try next member, but do not go to next group—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. Unified Communications Manager stops trying upon reaching the last member of the current line group.  
- Skip remaining members, and go directly to next group—If you choose this hunt option, Unified Communications Manager skips the remaining members of this line group when the RNA reversion timeout value elapses for the first member. Unified Communications Manager then proceeds directly to the next line group in a hunt list.  
- Stop hunting—If you choose this hunt option, Unified Communications Manager stops hunting after trying to distribute a call to the first member of this line group and the member does not answer the call. |
| Automatically Logout Hunt Member on No Answer | If this check box is checked, line members will be logged off the hunt list automatically. Line members can log back in using the "HLOG" softkey or PLK. |
For a given distribution algorithm, choose a hunt option for Unified Communications Manager to use if a call is distributed to a member of a line group that is busy. Choose from the options in the drop-down list box:

- **Try next member; then, try next group in Hunt List**—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. If unsuccessful, Unified Communications Manager then tries the next line group in a hunt list.

- **Try next member, but do not go to next group**—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. Unified Communications Manager stops trying upon reaching the last member of the current line group.

- **Skip remaining members, and go directly to next group**—If you choose this hunt option, Unified Communications Manager skips the remaining members of this line group upon encountering a busy member. Unified Communications Manager proceeds directly to the next line group in a hunt list.

- **Stop hunting**—If you choose this hunt option, Unified Communications Manager stops hunting after trying to distribute a call to the first busy member of this line group.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Busy</td>
<td>For a given distribution algorithm, choose a hunt option for Unified Communications Manager to use if a call is distributed to a member of a line group that is busy. Choose from the options in the drop-down list box:</td>
</tr>
<tr>
<td></td>
<td>• Try next member; then, try next group in Hunt List—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. If unsuccessful, Unified Communications Manager then tries the next line group in a hunt list.</td>
</tr>
<tr>
<td></td>
<td>• Try next member, but do not go to next group—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. Unified Communications Manager stops trying upon reaching the last member of the current line group.</td>
</tr>
<tr>
<td></td>
<td>• Skip remaining members, and go directly to next group—If you choose this hunt option, Unified Communications Manager skips the remaining members of this line group upon encountering a busy member. Unified Communications Manager proceeds directly to the next line group in a hunt list.</td>
</tr>
<tr>
<td></td>
<td>• Stop hunting—If you choose this hunt option, Unified Communications Manager stops hunting after trying to distribute a call to the first busy member of this line group.</td>
</tr>
</tbody>
</table>
For a given distribution algorithm, choose a hunt option for Unified Communications Manager to use if a call is distributed to a member of a line group that is not available. The Not Available condition occurs when none of the phones that are associated with the DN in question is registered. Not Available also occurs when extension mobility is in use and the DN/user is not logged in. Choose from the options in the drop-down list box:

- **Try next member; then, try next group in Hunt List**—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. If unsuccessful, Unified Communications Manager then tries the next line group in a hunt list.

- **Try next member, but do not go to next group**—If you choose this hunt option, Unified Communications Manager distributes a call to idle or available members starting from the first idle or available member of a line group to the last idle or available member. Unified Communications Manager stops trying upon reaching the last member of the current line group.

- **Skip remaining members, and go directly to next group**—If you choose this hunt option, Unified Communications Manager skips the remaining members of this line group upon encountering the first unavailable member. Unified Communications Manager proceeds directly to the next line group in a hunt list.

- **Stop hunting**—If you choose this hunt option, Unified Communications Manager stops hunting after trying to distribute a call to the first unavailable member of this line group.

**Line Group Member Information**

**Find Directory Numbers to Add to Line Group**
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Partition</td>
<td>Choose a route partition for this line group from the drop-down list box. The default value specifies &lt;None&gt;. If you click Find, the Available DN/Route Partition list box displays all DNs that belong to the chosen partition.</td>
</tr>
<tr>
<td>Directory Number Contains</td>
<td>Enter the character(s) that are found in the directory number that you are seeking and click the Find button. Directory numbers that match the character(s) that you entered display in the Available DN/Route Partition box.</td>
</tr>
<tr>
<td>Available DN/Route Partition</td>
<td>Choose a directory number in the Available DN/Route Partition list box and add it to the Selected DN/Route Partition list box by clicking Add to Line Group.</td>
</tr>
<tr>
<td>Current Line Group Members</td>
<td></td>
</tr>
<tr>
<td>Broadcast algorithm with shared line DNs</td>
<td>To change the priority of a directory number, choose a directory number in the Selected DN/Route Partition list box. Move the directory number up or down in the list by clicking the arrows on the right side of the list box. To reverse the priority order of the directory numbers in the Selected DN/Route Partition list box, click Reverse Order of Selected DNs/Route Partitions. <strong>Note</strong> When adding DNs and Route Partitions to your line group, do not put DNs that are shared lines in a line group that uses the Broadcast distribution algorithm. Unified Communications Manager cannot display all DNs that are shared lines on devices where the DNs are configured as shared lines if the DNs are members of a line group that uses the Broadcast distribution algorithm.</td>
</tr>
<tr>
<td>Removed DN/Route Partition</td>
<td>Choose a directory number in the Selected DN/Route Partition list box and add it to the Removed DN/Route Partition list box by clicking the down arrow between the two list boxes.</td>
</tr>
<tr>
<td>Directory Numbers</td>
<td></td>
</tr>
</tbody>
</table>
Add Members to Line Group

You can add members to a new line group or to an existing line group. The following procedure describes adding a member to an existing line group.

Before you begin
You must define one or more directory numbers before performing this procedure.

Procedure

Step 1  Choose Call Routing > Route/Hunt > Line Group.
Step 2  Locate the line group to which you want to add a member.
Step 3  If you need to locate a directory number, choose a route partition from the Partition drop-down list box, enter a search string in the Directory Number Contains field, and click Find. To find all directory numbers that belong to a partition, leave the Directory Number Contains field blank and click Find.

A list of matching directory numbers displays in the Available DN/Route Partition list box.

Step 4  In the Available DN/Route Partition list box, choose a directory number to add and click Add to Line Group to move it to the Selected DN/Route Partition list box. Repeat this step for each member that you want to add to this line group.

Step 5  In the Selected DN/Route Partition list box, choose the order in which the new directory number(s) is to be accessed in this line group. To change the order, click a directory number and use the Up and Down arrows to the right of the list box to change the order of directory numbers.

Step 6  Click Save to add the new directory numbers and to update the directory number order for this line group.

Remove Members From Line Group

You can remove members from a new line group or from an existing line group. The following procedure describes removing a directory number from an existing line group.

Procedure

Step 1  Choose Call Routing > Route/Hunt > Line Group.
Remove Members From Line Group

**Step 2**  Locate the line group from which you want to remove a directory number.

**Step 3**  In the Selected DN/Route Partition list box, choose a directory number to be deleted and click the down arrow below the list box to move the directory number to the Removed DN/Route Partition list box. Repeat this step for each member that you want to remove from this line group.

**Step 4**  To remove the members, click Save.
Configure Hunt Pilots

- Hunt Pilot Overview, on page 149
- Hunt Pilot Configuration Task Flow, on page 149
- Hunt Pilot Interactions and Restrictions, on page 154

Hunt Pilot Overview

A hunt pilot comprises a number or pattern and a set of associated digit manipulations that can route calls to a group of phones or directory numbers in a line group.

Hunt pilots work in conjunction with hunt lists, which are prioritized lists of eligible paths (line groups) for incoming calls. When a call is placed to a hunt pilot DN, the system offers the call to the first line group specified in the hunt list. If no one in the first line group answers the call, the system offers the call to the next line group specified in the hunt list. Line groups control the order in which the call is distributed to phones within the group. They point to specific extensions, which are typically IP phone extensions or voicemail ports. Line groups cannot point to Computer Telephony Integration (CTI) ports and CTI route points, so you cannot use hunt pilots to distribute calls to endpoints that are controlled through CTI applications such as Cisco Customer Response Solution (CRS) or IP Interactive Voice Response (IP IVR).

A hunt pilot can distribute calls to any of its assigned line groups, even if the line groups and the hunt pilot reside in different partitions. A call distributed by the hunt pilot overrides all the partitions and calling search space restrictions.

Hunt Pilot Configuration Task Flow

Complete these tasks to configure hunt pilots for your system. Hunt pilots can be used to route calls to a group of phones or directory numbers in a line group.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Configure Line Groups, on page 150</td>
<td>Create a line group to enable multiple phones to answer calls that are directed to a single directory number (DN).</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Hunt Lists, on page 150</td>
<td>Configure a hunt list with a prioritized order of line groups.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Hunt Pilots, on page 151</td>
<td>Configure a hunt pilot number or pattern that the system uses to direct calls to a hunt list.</td>
</tr>
</tbody>
</table>

Configure Line Groups

Line groups let multiple phones answer calls that are directed to a single directory number. The Distribution Algorithm controls the order in which an incoming call gets distributed to the phones in the group.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Line Group.

**Step 2**
Choose one of the following options:
- Click **Add New** to create a new line group.
- Click **Find** and select an existing line group.

**Step 3**
Enter a **Line Group Name**.

**Step 4**
From the **Distribution Algorithm** field, select the type of algorithm that you want to use to distribute calls.

**Step 5**
Configure the fields in the **Line Group Members to Add to Line Group** section to add directory numbers to the line group:
- a) Select a **Partition** where the directory numbers that you want to add reside.
- b) Optional. Filter the search by completing the **Directory Number Contains** field.
- c) Click **Find**. The list of Directory Numbers from the Partition appears in the box.
- d) In the **Available DN/Route Partition** list box, select each directory number that you want to add to the group and click **Add to Line Group**.

**Step 6**
Configure the remaining fields in the **Line Group Configuration** window. See the online help for more information about the fields and their configuration options.

**Step 7**
Click **Save**.

Configure Hunt Lists

A hunt list is a prioritized list of line groups. When the system routes a call through a hunt list, it uses the line groups in the order that you define in the hunt list.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Hunt List.

**Step 2**
Choose one of the following options:
- Click **Add New** to create a new list.
Configure Hunt Pilots

Configure a hunt pilot number or pattern that the system uses to route calls to a line group.

Note
For information about wildcards and special characters that you can use for the hunt pilot, see Wildcards and Special Characters in Hunt Pilots, on page 152.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Hunt Pilot.
Step 2 Choose one of the following options:
   • Click Add New to create a new hunt pilot.
   • Click Find and select an existing hunt pilot.
Step 3 In the Hunt Pilot field, enter the number or pattern that you want to use to route calls.
Step 4 From the Hunt List drop-down, select the hunt list to which you want to direct calls that match the hunt pilot number.
Step 5 Complete the remaining fields in the Hunt Pilot Configuration window. For help with the fields and their settings, see the online help.
Step 6 If you want to enable Call Queuing, check the Queue Calls check box and configure the fields in the Queuing section.
Step 7 Assign any digit transformation patterns that you want to apply to calling, connected or called parties.
Step 8 Click Save.
Wildcards and Special Characters in Hunt Pilots

Wildcards and special characters in hunt pilots allow a hunt pilot to match a range of numbers (addresses). Use these wildcards and special characters also to build instructions that enable the Cisco Unified Communications Manager to manipulate a number before sending it to an adjacent system.

The following table describes the wildcards and special characters that Cisco Unified Communications Manager supports.

**Table 16: Wildcards and Special Characters**

<table>
<thead>
<tr>
<th>Character</th>
<th>Description</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>@</td>
<td>The at symbol (@) wildcard matches all National Numbering Plan numbers. Each route pattern can have only one @ wildcard.</td>
<td>The route pattern 9.@ routes or blocks all numbers that the National Numbering Plan recognizes. The following route patterns examples show National Numbering Plan numbers that the @ wildcard encompasses: • 0 • 1411 • 19725551234 • 101028819725551234 • 01133123456789</td>
</tr>
<tr>
<td>X</td>
<td>The X wildcard matches any single digit in the range 0 through 9.</td>
<td>The route pattern 9XXX routes or blocks all numbers in the range 9000 through 9999.</td>
</tr>
<tr>
<td>!</td>
<td>The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.</td>
<td>The route pattern 91! routes or blocks all numbers in the range 910 through 91999999999999999999999.</td>
</tr>
<tr>
<td>?</td>
<td>The question mark (?) wildcard matches zero or more occurrences of the preceding digit or wildcard value.</td>
<td>The route pattern 91X? routes or blocks all numbers in the range 91 through 91999999999999999999999.</td>
</tr>
<tr>
<td>+</td>
<td>The plus sign (+) wildcard matches one or more occurrences of the preceding digit or wildcard value.</td>
<td>The route pattern 91X+ routes or blocks all numbers in the range 910 through 91999999999999999999999.</td>
</tr>
<tr>
<td>[ ]</td>
<td>The square bracket ([ ]) characters enclose a range of values.</td>
<td>The route pattern 813510[012345] routes or blocks all numbers in the range 8135100 through 8135105.</td>
</tr>
<tr>
<td>-</td>
<td>The hyphen (-) character, used with the square brackets, denotes a range of values.</td>
<td>The route pattern 813510[0-5] routes or blocks all numbers in the range 8135100 through 8135105.</td>
</tr>
<tr>
<td>Character</td>
<td>Description</td>
<td>Examples</td>
</tr>
<tr>
<td>-----------</td>
<td>-------------</td>
<td>----------</td>
</tr>
<tr>
<td>^</td>
<td>The circumflex (^) character, used with the square brackets, negates a range of values. Ensure that it is the first character following the opening bracket ([). Each route pattern can have only one ^ character.</td>
<td>The route pattern 813510[^0-5] routes or blocks all numbers in the range 8135106 through 8135109.</td>
</tr>
<tr>
<td>.</td>
<td>The dot (.) character, used as a delimiter, separates the Cisco Unified Communications Manager access code from the directory number. Use this special character, with the discard digits instructions, to strip off the Cisco Unified Communications Manager access code before sending the number to an adjacent system. Each route pattern can have only one dot (.) character.</td>
<td>The route pattern 9.@ identifies the initial 9 as the Cisco Unified Communications Manager access code in a National Numbering Plan call.</td>
</tr>
<tr>
<td>*</td>
<td>The asterisk (*) character can provide an extra digit for special dialed numbers.</td>
<td>You can configure the route pattern *411 to provide access to the internal operator for directory assistance.</td>
</tr>
<tr>
<td>#</td>
<td>The octothorpe (#) character generally identifies the end of the dialing sequence. Ensure the # character is the last character in the pattern.</td>
<td>The route pattern 901181910555# routes or blocks an international number that is dialed from within the National Numbering Plan. The # character after the last 5 identifies this digit as the last digit in the sequence.</td>
</tr>
<tr>
<td>+</td>
<td>A plus sign preceded by a backslash, that is, +, indicates that you want to configure the international escape character +.</td>
<td>Using + means that the international escape character + is used as a dialable digit, not as a wildcard.</td>
</tr>
</tbody>
</table>

### Performance and Scalability for Hunt Pilots

The following performance and scalability restrictions apply:

- A single Unified CM Cluster supports a maximum of 15,000 hunt list devices.
- A single Unified CM Subscriber supports a maximum of 100 hunt pilots with call queuing enabled per node
- Hunt list devices may be a combination of 1500 hunt lists with ten IP phones in each hunt list, 750 hunt lists with twenty IP phones in each hunt list, or similar combinations.
When using the broadcast algorithm for call coverage, the number of hunt list devices is limited by the number of busy hour call attempts (BHCA). Note that a BHCA of 10 on a hunt pilot pointing to a hunt list or hunt group containing 10 phones and using the broadcast algorithm is equivalent to 10 phones with a BHCA of 10.

- The maximum number of simultaneous callers in queue for each hunt pilot that you can configure ranges from 1 to 100 (default 32).

- The maximum wait time in queue for each hunt pilot that you can configure ranges from 0 to 3600 seconds (default 900). An increase in the number of hunt lists can require you to increase the dial plan initialization timer that is specified in the Unified Communications Manager service parameters. We recommend that you set the dial plan initialization timer to 600 seconds if you have 1500 hunt lists configured.

- We recommend having no more than 35 directory numbers for a single line group when using broadcast algorithms with call queuing. Additionally, the number of broadcast line groups depends on the busy hour call completion rate (BHCC). If there are multiple broadcast line groups in a Unified CM system, the number of maximum directory numbers in a line group must be less than 35. The number of busy hour call attempts (BHCA) for all the broadcast line groups should not exceed 35 calls set up per second.

### Hunt Pilot Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions and Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Single Number Reach with Hunt Groups</td>
<td>If you have a hunt group configured and one or more of the directory numbers that the hunt group points toward also has Single Number Reach (SNR) enabled, the call does not extend to the SNR remote destinations unless all devices in the hunt group are logged in. For each device within the hunt group, the <strong>Logged Into Hunt Group</strong> check box must be checked within the <strong>Phone Configuration</strong> window for that device.</td>
</tr>
<tr>
<td>Call Queuing</td>
<td>Call Queuing is a subfeature of hunt pilots. When call queuing is enabled and the incoming call requirement to a particular hunt pilot exceeds the number of hunt members whom are available to answer a call, the system queues incoming calls until a hunt member is available to answer them. You can configure announcements and music on hold to play to callers while they are waiting. For additional configuration details, see the <strong>Configure Call Queuing</strong> chapter of the <strong>Feature Configuration Guide for Cisco Unified Communications Manager</strong>.</td>
</tr>
</tbody>
</table>
## Calls Not Being Distributed

*Table 17: Calls are not being distributed with circular algorithm*

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calls are not being distributed correctly in Circular algorithm for a line group with BOT and TCT devices.</td>
<td>When a call is extended to an agent who is in a logged off state and the call is rejected with a different reject type other than the &quot;Huntlogout&quot; type. Then the index will not get incremented and the call will go to the same agent who had answered the previous call.</td>
</tr>
<tr>
<td>Calls are not distributed correctly in Circular algorithm for a line group.</td>
<td>While distributing the calls in a circular algorithm, when an agent is busy, the call is extended to the next available agent (i.e. the next agent will answer the call on behalf of the busy agent).</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>In the case of multiple calls at the same time, the next available agent answers the call.</td>
</tr>
</tbody>
</table>
CHAPTER 22

Configure Translation Patterns

• Translation Pattern Overview, on page 157
• Translation Pattern Prerequisites, on page 157
• Translation Pattern Configuration Task Flow, on page 158

Translation Pattern Overview

You can configure translation patterns to manipulate digits for any type of call. Translation patterns follow the same general rules and use the same wildcards as route patterns. As with route patterns, you assign a translation pattern to a partition. However, when the dialed digits match the translation pattern, Cisco Unified Communications Manager does not route the call to an outside entity such as a gateway; instead, it performs the translation first and then routes the call again, this time using the calling search space configured within the translation pattern.

Translation Pattern Prerequisites

Before you configure a translation pattern, you must complete the following tasks:

• Partition Configuration Task Flow, on page 116
• Call Routing Configuration Task Flow, on page 126

Note

For each translation pattern that you create, ensure that the combination of partition, route filter, and numbering plan is unique. If you receive an error that indicates duplicate entries, check the route pattern or hunt pilot, translation pattern, directory number, call park number, call pickup number, or meet-me number configuration windows.
Translation Pattern Configuration Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Translation Patterns, on page 158</td>
<td>Configure translation patterns to specify how to route a call after it is placed.</td>
</tr>
</tbody>
</table>

**Configure Translation Patterns**

Configure translation patterns to apply digit manipulations to the calling and called numbers when the dial string matches the pattern. The system completes the digit translation and then reroutes the call.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Call Routing > Translation Pattern**.

**Step 2** Choose one of the following options:

- Click **Add New** to add a new translation pattern.
- Click **Find**, and select an existing translation pattern.

**Step 3** In the **Translation Pattern** field, enter the pattern that you want the system to match to dial strings that use this pattern.

**Step 4** From the **Partition** drop-down list, select the partition where you want to assign this pattern.

**Step 5** Complete the remaining fields in the **Translation Pattern Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 6** Click **Save**.
CHAPTER 23

Configure Transformation Patterns

- Transformation Pattern Overview, on page 159
- Transformation Pattern Configuration Task Flow, on page 159

Transformation Pattern Overview

Transformation patterns determine how the system manipulates the digits that were dialed in an incoming or outgoing call. You can configure transformation patterns when you need to change the calling or called number before the system sends it to the phone or the PSTN.

You can use transformation patterns to discard digits, prefix digits, add a calling party transformation mask, and control the presentation of the calling party number.

You can:
  - Hit a Calling Party Transformation Pattern with a Called Party Transformation CSS.
  - Hit a Called Party Transformation Pattern with a Calling Party Transformation CSS.

Transformation Pattern Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Calling Party Transformation Patterns, on page 160</td>
<td>Use this procedure to transform the calling number. For example, you can configure a transformation pattern that replaces a caller's extension with the office's main number when calling the PSTN.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Called Party Transformation Patterns, on page 160</td>
<td>Use this procedure to transform the called number. For example, you can configure a transformation pattern that retains only the last five digits of a call dialed as a ten-digit number.</td>
</tr>
</tbody>
</table>
Configure Calling Party Transformation Patterns

Use this procedure to transform the calling number. For example, you can configure a transformation pattern that replaces a caller's extension with the office's main number when calling the PSTN.

Procedure

Step 1
From Cisco Unified CM Administration, choose Call Routing > Transformation > Transformation Pattern > Calling Party Transformation Pattern.

Step 2
Choose one of the following options:
- Click Add New to add a new calling party transformation pattern.
- Click Find and select an existing pattern.

Step 3
From the Pattern field, enter the pattern that you want to match to the calling party number.

Note

For Outbound Calls:

The calling party transformation mask is selected based on the pre transform calling party number (extension assigned to the IP Phone).

While selecting the calling party transformation mask on the SIP trunk, if the calling party number is transformed to a different number on either the route pattern/group, the pre transform calling number is always used to select the calling party transformation mask.

Whereas according to the Dialed Number Analyzer (DNA), the transformed number is used to select the calling party transformation mask. However, this is the wrong behavior of DNA.

Step 4
Complete the remaining fields in the Calling Party Transformation Pattern Configuration window. For more information on the fields and their configuration options, see Online Help.

Step 5
Click Save.

Configure Called Party Transformation Patterns

Use this procedure to transform the called number. For example, you can configure a transformation pattern that retains only the last five digits of a call dialed as a ten-digit number.
Configure the Dial Plan

## Configure Transformation Profiles

Perform this procedure only if you are using Cisco Intercompany Media Engine (Cisco IME). You must configure a transformation profile to convert dialed numbers into the E.164 format. The E.164 format includes the international "+" prefix; for example, "+14085551212".

### Procedure

**Step 1**  
From Cisco Unified CM Administration, choose **Call Routing > Transformation > Transformation Profile**.

**Step 2**  
Choose one of the following options:
- Click **Add New**, to add a new transformation profile.
- Click **Find** and choose a pattern from the resulting list, to modify the settings for an existing transformation profile.

The **Transformation Profile Configuration** window appears.

**Step 3**  
Configure the fields in the **Transformation Profile Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 4**  
Click **Save**.
Configure Dial Rules

- **Dial Rules Overview, on page 163**
- **Dial Rules Prerequisites, on page 163**
- **Dial Rules Configuration Task Flow, on page 164**
- **Dial Rules Interactions and Restrictions, on page 169**

### Dial Rules Overview

The Unified CM supports the following types of dial rules:

- **Application Dial Rules**: The administrator uses application dial rules to add and sort the priority of dialing rules for applications such as Cisco web dialer and Cisco Unified Communications Manager Assistant.

- **Directory Lookup Dial Rules**: The administrator uses directory lookup dial rules to transform caller identification numbers and perform a directory search from the assistant console in application such as Cisco Unified Communications Manager Assistant.

- **SIP Dial Rules**: The administrator uses SIP dial rules to perform system digit analysis and routing. The administrator configures SIP dial rules and adds the SIP dial rule to the Cisco Unified IP Phone before the call processing takes place.

### Dial Rules Prerequisites

- For SIP dial rules configuration, the devices must be running SIP

- The administrator associates the SIP dial rules with the following devices: Cisco IP Phones 7911, 7940, 7941, 7960, 7961, 7970, and 7971
## Dial Rules Configuration Task Flow

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Application Dial Rules, on page 164</td>
<td>Configure application dial rules to add and sort the priority of dialing rules for applications such as Cisco web dialer and Cisco Unified Communications Manager Assistant.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Directory Lookup Dial Rules, on page 165</td>
<td>Configure directory lookup dial rules to transform caller identification numbers into numbers that can be looked up in the directory.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure SIP Dial Rules, on page 165</td>
<td>Use SIP dial rules configuration to configure dial plans for phones that are running SIP.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Reprioritize Dial Rule, on page 168</td>
<td>Optional. Change the priority of the dial rules in the Cisco Unified Communications Manager Administration window, if more than one dial rule exists.</td>
</tr>
</tbody>
</table>

### Configure Application Dial Rules

Cisco Unified Communications Manager supports application dial rules that allow you to add and sort the priority of dialing rules for applications such as Cisco web dialer and Cisco Unified Communications Manager Assistant. Application dial rules automatically strip numbers from or add numbers to telephone numbers that the user dials. For example, the dial rules automatically add the digit 9 in front of a 7-digit telephone number to provide access to an outside line.

---

**Note**

Cisco Unified Communications Manager automatically applies application dial rules to all remote destination numbers for CTI remote devices.

Perform the following procedure to add a new application dial rule or update an existing application dial rule.

### Procedure

1. **Step 1** From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Rules > Application Dial Rules.
2. **Step 2** In the Find and List Application Dial Rules window, perform one of the following steps:
   - Click Add New.
   - Click Find and choose an existing application dial rule.
3. **Step 3** Configure the fields in the Application Dial Rule Configuration window. For detailed field descriptions, refer to the online help.
Configure the Dial Plan

Configure Directory Lookup Dial Rules

Step 4
Click Save.

What to do next
Perform the following tasks:

- Configure Directory Lookup Dial Rules, on page 165
- Configure SIP Dial Rules, on page 165

Configure Directory Lookup Dial Rules

Directory lookup dial rules transform caller identification numbers into numbers that can be looked up in the directory. Each rule specifies which numbers to transform, based on the beginning digits and length of the number. For example, you can create a directory lookup dial rule that automatically removes the area code and two prefix digits from a 10-digit telephone, which would transform 4085551212 into 51212.

Perform the following procedure to add a new directory lookup dial rule or update an existing directory lookup dial rule.

Procedure

Step 1
From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Rules > Directory Lookup Dial Rules.

Step 2
In the Directory Lookup Dial Rule Find and List window, perform one of the following steps:

- Click Add New.
- Click Find and choose an existing directory lookup dial rule.

Step 3
Configure the fields in the Directory Lookup Dial Rule Configuration window. For detailed field descriptions, refer to the online help.

Step 4
Click Save.

What to do next
Configure SIP Dial Rules, on page 165

Configure SIP Dial Rules

SIP dial rules provide local dial plans for Cisco IP Phones that are running SIP, so users do not have to press a key or wait for a timer before the call gets processed. The administrator configures the SIP dial rule and applies it to the phone that is running SIP.
Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Set Up SIP Dial Rule, on page 167</td>
<td>Configure and update SIP dial rules and associate them with the phones that are running SIP.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Reset SIP Dial Rule, on page 167</td>
<td>Reset or restart the phone that is running SIP when the SIP dial rule gets updated, so that the phone is updated with the new SIP dial rule.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Synchronize SIP Dial Rules Settings With SIP Phones, on page 168</td>
<td>(Optional) Synchronize a SIP phone with a SIP dial rule that has undergone configuration changes, which applies any outstanding configuration settings in the least intrusive manner possible. For example, a reset or restart may not be required on some affected SIP phones.</td>
</tr>
</tbody>
</table>

Related Topics

- Pattern Formats, on page 166

Pattern Formats

Table 18: Pattern Formats for SIP Dial Rules

<table>
<thead>
<tr>
<th>Dial Rule Pattern</th>
<th>Value</th>
</tr>
</thead>
</table>
| 7940_7960_OTHER   | • Period (.) matches any character  
 • Pound sign (#) acts as the terminating key, and you can apply termination only after matching hits. Alternatively asterisk (*) can also be used as a terminating key as well.  
 **Note** You must configure the pound sign in the pattern field so that it is valid for 7940_7960_OTHER.  
 • Asterisk (*) matches one or more characters and it gets processed as a wildcard character. You can override this by preceding the * with a backward slash (\) escape sequence, which results in the sequence \\*. The phone automatically strips the \, so it does not appear in the outgoing dial string. When * is received as a dial digit, it gets matched by the wildcard characters * and period (.).  
 • Comma (,) causes the phone to generate a secondary dial tone. For example, 7..., will match any 4-digit DN that starts with 7. 8,..., will match 8, play secondary dial tone (default value), and then match any 5-digit DN. |
Set Up SIP Dial Rule

To configure dial plans for phones that are running SIP.

Procedure

Step 1  From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Rules > SIP Dial Rules.

Step 2  In the Find and List SIP Dial Rules window. Perform one of the following steps:

- Click Add New
- Click Find and choose an existing SIP Dial Rule

Step 3  Configure the fields in the SIP Dial Rule Configuration window. For detailed field descriptions, refer to the online help.

Step 4  Click Save.

Note  When you add or update a SIP dial rule in Cisco Unified Communications Manager Administration, be aware that the Cisco TFTP service rebuilds all phone configuration files, which may cause CPU to spike on the server where the Cisco TFTP service runs, especially if you have a large system with many phones. To ensure that CPU does not spike, add or update the SIP dial rule during a maintenance window or temporarily stop the Cisco TFTP service in Cisco Unified Serviceability before you make the configuration change. If you stop the Cisco TFTP service, remember to restart the service in Cisco Unified Serviceability after you add or update the SIP dial rule.

What to do next

Reset SIP Dial Rule, on page 167

Related Topics

Pattern Formats, on page 166

Reset SIP Dial Rule

Perform the following procedure to reset or restart the phone that is running SIP when the SIP dial rule gets updated, so the phone gets updated with the new SIP dial rule.

Before you begin

Set Up SIP Dial Rule, on page 167

Procedure

Step 1  From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Rules > Application Dial Rules.

Step 2  In the Find and List SIP Dial Rules window, click Find and choose an existing SIP dial rule that you want to reset.
Step 3 In the SIP Dial Rule Configuration window, click Reset.

Step 4 Perform one of the following tasks in the Device Reset dialog box:

- To restart the chosen devices without shutting them down and reregister them with Cisco Unified Communications Manager, click Restart.

- To shut down, and then restart the device, click Reset.

- To close the Device Reset dialog box without performing any action, click Close.

After the administrator configures the SIP dial rule and applies it to the phone that is running SIP, the database sends the TFTP server a notification, so it can build a new set of configuration files for the phone that is running SIP. The TFTP server notifies Cisco Unified Communications Manager about the new configuration file, and the updated configuration file is sent to the phone. See Configure TFTP Servers for Cisco Unified IP phones that run SIP for more information.

---

What to do next

Synchronize SIP Dial Rules Settings With SIP Phones, on page 168

Synchronize SIP Dial Rules Settings With SIP Phones

To synchronize a SIP phone with a SIP dial rule that has undergone configuration changes, perform the following procedure.

Before you begin

Reset SIP Dial Rule, on page 167

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Rules > SIP Dial Rules.

Step 2 In the Find and List SIP Dial Rules window, click Find and choose an existing SIP dial rule to which you want to synchronize applicable SIP phones.

Step 3 Make any additional configuration changes and click Save in the SIP Dial Rule Configuration.

Step 4 Click Apply Config.

Step 5 Click OK.

---

Reprioritize Dial Rule

To add and sort the priority of dialing rules in the Dial Rule Configuration window.

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose Call Routing > Dial Rules.
Step 2  Select one of the following:

- Application Dial Rules
- Directory Lookup Dial Rules
- SIP Dial Rules

Step 3  In the Find and List window, choose a dial rule and click the dial rule name. The Dial Rule Configuration window appears.

Step 4  Use the up and down arrows to move the dial rule up or down the list.

Step 5  After you complete prioritizing the order, click Save.

---

Dial Rules Interactions and Restrictions

SIP Dial Rules Interactions

<table>
<thead>
<tr>
<th>Cisco Unified IP Phone</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>7911, 7941, 7961, 7970, and 7971 that are running SIP</td>
<td>These phones use the 7940_7960_OTHER dial rules patterns. Key Press Markup Language (KPML) allows for the digits to be sent to Cisco Unified Communications Manager digit by digit; SIP dial rules allow for a pattern of digits to be collected locally on the phone prior to sending to Cisco Unified Communications Manager. If SIP dial rules are not configured, KPML is used. To increase the performance of Cisco Unified Communications Manager (increasing the number of calls that get processed), Cisco recommends that administrators configure SIP dial rules.</td>
</tr>
<tr>
<td>7940 and 7960 that are running SIP</td>
<td>These phones use the 7940_7960_OTHER dial rules pattern and do not support KPML. If the administrator does not configure a SIP dial plan for these phones, the user must wait a specified time before the digits are sent to Cisco Unified Communications Manager for processing. This delays the processing of the actual call.</td>
</tr>
</tbody>
</table>
Directory Lookup Dial Rules Restrictions

<table>
<thead>
<tr>
<th>Field</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Begins With</td>
<td>This field supports only digits and the characters +, *, and #. The length cannot exceed 100 characters.</td>
</tr>
<tr>
<td>Number of Digits</td>
<td>This field supports only digits, and the value in this field cannot be less than the length of the pattern that is specified in the pattern field.</td>
</tr>
<tr>
<td>Total Digits to be Removed</td>
<td>This field supports only digits, and the value in this field cannot be more than the value in the Number of Digits field.</td>
</tr>
<tr>
<td>Prefix with Pattern</td>
<td>The prefix it with field supports only digits and the characters +, *, and #. The length cannot exceed 100 characters. Note: You cannot allow both the Total Digits to be Removed field and the Prefix with Pattern field to be blank for a dial rule.</td>
</tr>
</tbody>
</table>
Intercluster Lookup Service Overview

The Intercluster Lookup Service (ILS) allows you to create networks of remote Cisco Unified Communications Manager clusters. When you configure ILS on multiple clusters, it updates Cisco Unified Communications Manager with the current status of remote clusters in the ILS network.

In Cisco Unified CM Administration, you can configure ILS on a pair of clusters and then join those clusters to form an ILS network. ILS allows you to join additional clusters to the network without having to configure the connections between each cluster.

An ILS network comprises the following components:

- Hub clusters
- Spoke clusters
- Global dial plan imported catalogs

Hub Clusters

Hub clusters form the backbone of an ILS network. Hub clusters exchange ILS updates with the other hub clusters in the ILS network, and then relay that information to and from their spoke clusters.

When a new hub cluster registers to another hub cluster in an existing ILS network, ILS automatically creates a full mesh connection between the new hub cluster and all the existing hub clusters in the ILS network.

Spoke Clusters

A spoke cluster connects to the hub cluster in an ILS network to relay ILS updates to and from the rest of the ILS network. Spoke clusters contact only their local hub cluster and never directly contact other hub clusters or other spoke clusters.
Global Dial Plan Imported Catalogs

To provide URI dialing compatibility with third-party systems, you can manually import a third-party directory URI or +E.164 number catalog from a CSV file into any hub cluster in the ILS network. ILS maintains the imported catalog and replicates that catalog out to the other clusters in the network. You can dial one of the third-party directory URIs or +E.164 numbers catalog from any server in the ILS network.

ILS Prerequisites

You must study your network and design an ILS topology.


ILS Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Activate Intercluster Lookup Service, on page 173</td>
<td>Activate Intercluster Lookup Service to configure cluster IDs and remote clusters.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Cluster IDs, on page 173</td>
<td>Provide a unique identifier for each cluster in the ILS network.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Remote Clusters, on page 174</td>
<td>Configure remote clusters in the ILS network.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>To activate ILS on the various clusters, complete the following tasks:</td>
<td>Activate ILS on the hub cluster and spoke cluster in the ILS network.</td>
</tr>
<tr>
<td></td>
<td>• Activate ILS on the Hub Cluster, on page 175</td>
<td>Note You must configure each cluster in your ILS network as either a hub cluster or a spoke cluster.</td>
</tr>
<tr>
<td></td>
<td>• Activate ILS on the Spoke Cluster, on page 175</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>(Optional) Configure authentication between your cluster. Select one of the following procedures:</td>
<td>Use TLS authentication between clusters in the ILS network.</td>
</tr>
<tr>
<td></td>
<td>• Enable TLS Authentication Between Clusters, on page 176</td>
<td>Use password authentication between remote clusters in the ILS network.</td>
</tr>
<tr>
<td></td>
<td>• Enable Password Authentication Between Clusters, on page 177</td>
<td>Use TLS and password authentication to setup a ILS network using common Certificate Authority (CA) signed certificates without exchanging self-signed certificates between clusters.</td>
</tr>
<tr>
<td></td>
<td>• Enable TLS with Password Authentication Between Clusters, on page 177</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 6</strong></td>
<td>Enable ILS Support for Global Dial Plan Replication, on page 187</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Import Catalogs in ILS Network, on page 178</td>
</tr>
</tbody>
</table>

(Optional) Enable ILS support for Global Dial Plan Replication to share dial plan information between participating ILS enabled clusters.

(Optional) To provide URI dialing compatibility with third party systems, you can manually import a third party directory URI or +E.164 number catalog from a csv file into any hub cluster in the ILS network.

---

### Activate Intercluster Lookup Service

You must activate the Intercluster Lookup Service to configure Cluster IDs and Remote Clusters.

**Procedure**

**Step 1**
From Cisco Unified Serviceability, choose **Tools > Service Activation**.

**Step 2**
From the **Server** drop-down list, choose the node on which you want to activate Cisco Intercluster Lookup Service, and then click **Go**.

**Step 3**
Check the **Cisco Intercluster Lookup Service** check box.

**Step 4**
Click **Save**.

---

**What to do next**

Configure Cluster IDs, on page 173

---

### Configure Cluster IDs

You must configure a unique cluster ID for each cluster in the ILS network. You must also ensure that you have a unique peer ID. The clusters use this unique cluster ID and peer ID when they exchange status messages.

For example, if you have an existing ILS network of four Cisco Unified Communications Manager clusters and you want to add an additional cluster, you can configure ILS on the new cluster and then register that cluster to any hub cluster in the existing ILS network. ILS automatically informs the new cluster of all clusters in the existing network.

Each cluster in an ILS network exchange and update messages, called peer info vectors, that are designed to inform remote clusters of the status of each cluster in the network. The update messages contain information about the known clusters in the network, including:

- Cluster IDs
- Peer IDs for the publisher
- Cluster descriptions and versions
- Fully Qualified Domain Name (FQDN) of the host
• IP addresses and host names for the cluster nodes that have ILS activated

Perform the following procedure to configure a unique identifier for each cluster in the network.

**Before you begin**

*Activate Intercluster Lookup Service, on page 173*

**Procedure**

**Step 1** Log into the Unified Communications Manager publisher node.

**Step 2** In Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.

**Step 3** In the **Enterprise Parameters Configuration** window **Cluster ID** field, enter a name of the cluster that you want to configure in your network.

You can enter up to 50 characters. You can enter alphanumeric characters, period (.), and hyphen (-). The default value is **StandAloneCluster**.

**Step 4** Click **Save**.

**What to do next**

*Configure Remote Clusters, on page 174*

---

**Configure Remote Clusters**

Perform the following steps to configure remote clusters in the ILS network.

**Before you begin**

*Configure Cluster IDs, on page 173*

**Procedure**

**Step 1** In Cisco Unified Communications Manager Administration, choose **Advanced Features > Cluster View**.

**Step 2** In the **Find and List Remote Clusters** window, choose any previously created remote cluster.

**Step 3** From the **Remote Cluster Service Configuration** window, check the appropriate check box to configure services such as Extension Mobility Cross Cluster, TFTP, and RSVP Agent for remote clusters.

**What to do next**

Perform one of the following procedures:

• **Activate ILS on the Hub Cluster, on page 175**

• **Activate ILS on the Spoke Cluster, on page 175**
Activate ILS on the Hub Cluster

You must configure each cluster in your ILS network as either a hub cluster or a spoke cluster. Each ILS network must have at least one hub cluster. You can connect a hub cluster to other hub clusters, or you can configure a hub cluster as the only hub cluster in the network. In addition, you can connect a hub cluster to multiple spoke clusters, or you can configure the hub cluster with no spoke clusters.

Perform the following procedure to activate the ILS on the hub cluster in the ILS network.

Before you begin
Configure Remote Clusters, on page 174

Procedure

Step 1 Log in to the Cisco Unified Communications Manager publisher node.
Step 2 Choose Advanced Features > ILS Configuration.
Step 3 In the ILS Configuration window, from the Role drop-down list, select Hub Cluster and click Save.
Step 4 In the ILS Configuration Registration popup window, leave the Registration Server text box empty and click OK.

What to do next
• Activate ILS on the Spoke Cluster, on page 175

Activate ILS on the Spoke Cluster

A spoke cluster connects to the hub cluster in an ILS network to relay ILS updates to and from the rest of the ILS network. Follow this procedure to activate ILS on the spoke cluster.

Before you begin
• Configure Cluster IDs, on page 173
• Configure Remote Clusters, on page 174

Procedure

Step 1 Log in to the Unified Communications Manager publisher node.
Step 2 In Cisco Unified CM Administration, choose Advanced Features > ILS Configuration.
Step 3 From the Role drop-down list, select Spoke Cluster and click Save.
Step 4 In the ILS Configuration Registration popup window, enter the IP address or fully qualified domain name of the publisher node for an existing hub cluster in your ILS network in the Registration Server text box and click OK.
Step 5 Confirm that your ILS network is configured by viewing the network in the ILS Clusters and Global Dial Plan Imported Catalogs section.
When the full network appears, your ILS network is configured for cluster discovery.

What to do next
Perform any of these optional procedures:

• Enable TLS with Password Authentication Between Clusters, on page 177
• Enable TLS Authentication Between Clusters, on page 176
• Enable Password Authentication Between Clusters, on page 177
• Enable ILS Support for Global Dial Plan Replication, on page 187

Enable TLS Authentication Between Clusters

(Optional) Use this procedure for the TLS authentication to encrypt communications between remote clusters in the ILS network:

Before you begin
To use Transport Layer Security (TLS) authentication between clusters, you must exchange Tomcat certificates between the publisher node of each cluster in the ILS network. From Cisco Unified Operating System Administration, use the Bulk Certificate Management feature to:

• export certificates from the publisher node to a central location, for each cluster in your network
• consolidate exported certificates from any publisher node server in your ILS network
• import certificates into the publisher node for each cluster in your network

Note

Procedure

Step 1
Log in to the Unified Communications Manager publisher node.

Step 2
In Cisco Unified CM Administration, choose Advanced Features > ILS Configuration.

Step 3
In the ILS Configuration window, check the Use TLS Certificates check box under ILS Authentication.

Step 4
Click Save.

What to do next
Perform any of these optional procedures:

• Enable Password Authentication Between Clusters, on page 177
• Enable ILS Support for Global Dial Plan Replication, on page 187
Enable Password Authentication Between Clusters

(Optional) To use password authentication between remote clusters, you must assign a password for all communications between clusters in your ILS network.

**Procedure**

1. Log in to the Unified Communications Manager publisher node.
2. In Cisco Unified CM Administration, choose Advanced Features > ILS Configuration.
3. In the ILS Configuration window, check the Use Password check box under ILS Authentication.
4. Enter a password in the Use Password text box.
   **Note**  You must configure all clusters in your network with the same password.
5. Re-enter the password in the Confirm Password text box.
6. Click Save.

**What to do next**

Perform any of these optional procedures:

- Enable TLS Authentication Between Clusters, on page 176
- Enable ILS Support for Global Dial Plan Replication, on page 187

Enable TLS with Password Authentication Between Clusters

**Before you begin**

To use Transport Layer Security (TLS) and password authentication without exchanging certificates between clusters, you must upload the certificate authority root certificates to the Tomcat trust and get the Tomcat certificate signed by the certificate authority root certificate. The certificate is then imported back on the same cluster. The clusters can be connected to Intercluster Lookup Service (ILS) network once the certificates are uploaded with the same password for all the clusters.


**Procedure**

1. Log in to the Cisco Unified Communications Manager publisher node.
2. In Cisco Unified CM Administration, choose Advanced Features > ILS Configuration.
3. In the ILS Configuration window, check the Use TLS Certificates check box under ILS Authentication.
Step 4  
In the **ILS Configuration** window, check the **Use Password** check box under ILS Authentication.

Step 5  
Enter a password in the **Use Password** text box.

**Note**  
You must configure all clusters in your network with the same password.

Step 6  
Re-enter the password in the **Confirm Password** text box.

Step 7  
Click **Save**.

---

**What to do next**

(Optional) **Enable ILS Support for Global Dial Plan Replication, on page 187**

---

**Enable ILS Support for Global Dial Plan Replication**

(Optional) To enable ILS support for Global Dial Plan Replication in the local cluster, follow this procedure:

**Procedure**

---

Step 1  
Log in to the Unified Communications Manager publisher node.

Step 2  
In Cisco Unified Communications Manager Administration, choose **Advanced Features > ILS Configuration**.

Step 3  
In the **ILS Configuration** window, check the **Exchange Global Dial Plan Replication Data with Remote Clusters** check box.

Step 4  
In the **Advertised Route String** text box, enter a route string for the local cluster.

Step 5  
Click **Save**.

**Note**  
When advertising URI patterns (**user@domain**), in the **SIP Profile Configuration** window, make sure that the **Dial String Interpretation** field is set to **Always treat all dial strings as URI addresses** to prevent the devices to dial URI learned patterns with only numbers in the user section as Directory Number patterns. Alternatively, you can advertise only URI patterns with text strings in the user section through ILS.

---

**What to do next**

**Import Catalogs in ILS Network, on page 178**

---

**Import Catalogs in ILS Network**

(Optional) To provide URI dialing compatibility with third party systems, you can manually import a third party directory URI or +E.164 number catalog from a csv file into any hub cluster in the ILS network. To Import Catalogs in the ILS network, follow this procedure:
Procedure

Step 1  In Cisco Unified Communications Manager Administration, choose Call Routing > Global Dial Plan Replication > Imported Global Dial Plan Catalogs.

Step 2  In the Find and List Imported Global Dial Plan Catalogs window, click Add New.

Step 3  Enter a Name, Description and Route String for the catalog and click Save.

Step 4  In Cisco Unified Communications Manager Administration, choose Bulk Administration > Upload/Download Files.

Step 5  Click Choose and select the CSV file that you want to import for the catalogs.

Step 6  From the Select the Target drop-down list, choose Imported Directory URIs and Patterns.

Step 7  From the Select Transaction Type drop-down list, choose Insert Imported Directory URIs and Patterns.

Step 8  Click Save.

ILS Interactions and Restrictions

ILS Interactions

Table 19: ILS Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
</table>
| Cluster discovery | ILS cluster discovery allows Cisco Unified Communications Manager clusters to learn dynamically about remote clusters without the need for an administrator to manually configure connections between those clusters. Each cluster in an ILS network exchange update messages, called peer info vectors, that are designed to inform remote clusters of the status of each cluster in the network. The update messages contain information about the known clusters in the network, including:  
  • Cluster IDs  
  • Cluster descriptions and versions  
  • Fully qualified domain name of the host  
  • IP addresses and hostnames for the cluster nodes that have ILS activated  

The ILS cluster discovery feature automatically populates the list of remote clusters that can be viewed in Cisco Unified CM Administration by choosing Advanced Features > Cluster View. From this window, you can configure services such as Extension Mobility Cross Cluster, TFTP, and RSVP Agent for remote clusters.  

**Note**  A fully qualified domain name of the remote cluster, as seen in the Cluster View, must be DNS resolvable for ILS discovery to work.
When Global Dial Plan Replication is enabled across an ILS network, remote clusters in an ILS network share global dial plan data, including the following:

- Directory URIs
- Alternate numbers
- Alternate number patterns
- Route strings
- PSTN failover numbers

To block Inbound calls based on calling party number in an ILS-based network, you must include the SIP route pattern's partition in the calling party's CSS. For example, if the call originates from SIP Trunk then SIP trunk inbound CSS must have SIP route pattern's partition.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Global Dial Plan Replication</td>
<td>When Global Dial Plan Replication is enabled across an ILS network, remote clusters in an ILS network share global dial plan data, including the following:</td>
</tr>
<tr>
<td></td>
<td>• Directory URIs</td>
</tr>
<tr>
<td></td>
<td>• Alternate numbers</td>
</tr>
<tr>
<td></td>
<td>• Alternate number patterns</td>
</tr>
<tr>
<td></td>
<td>• Route strings</td>
</tr>
<tr>
<td></td>
<td>• PSTN failover numbers</td>
</tr>
<tr>
<td>Block Inbound Calls</td>
<td>To block Inbound calls based on calling party number in an ILS-based network, you must include the SIP route pattern's partition in the calling party's CSS. For example, if the call originates from SIP Trunk then SIP trunk inbound CSS must have SIP route pattern's partition.</td>
</tr>
</tbody>
</table>

**Table 20: ILS Restrictions**

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ILS Service</td>
<td>The ILS Service runs only on the Unified Communications manager publisher node.</td>
</tr>
<tr>
<td>Clusters</td>
<td>A hub cluster can have many spokes but, a spoke cluster can have only one hub cluster.</td>
</tr>
<tr>
<td>ILS Network</td>
<td>You cannot connect a third-party call control system into an ILS network.</td>
</tr>
<tr>
<td>Cluster Import</td>
<td>You can import a third-party catalog into a hub cluster only.</td>
</tr>
<tr>
<td>Duplicated URI</td>
<td>If a learned ILS cluster contains duplicated URIs from a different remote cluster and when a call is placed to that URI, it will be routed to the cluster whose URI has been learned and inserted into the database first.</td>
</tr>
<tr>
<td>Database Replication Status</td>
<td>Although the Global dial plan data is exchanged successfully on the ILS Network, an ILS receiving cluster will not write learned information into the database until it completes its database replication status.</td>
</tr>
<tr>
<td>Import</td>
<td>For imported third-party directory URIs and patterns, the CSV file format must match the exact syntax as shown in the administration window sample file otherwise, the import fails.</td>
</tr>
<tr>
<td>Description</td>
<td>Restriction</td>
</tr>
<tr>
<td>-------------</td>
<td>-------------</td>
</tr>
<tr>
<td>When adding an additional hub cluster into the ILS network ensure to verify the following conditions are met for the primary ILS hub node:</td>
<td>ILS Hub</td>
</tr>
<tr>
<td>• Cluster ID is unique across all the hub nodes in the ILS cluster.</td>
<td></td>
</tr>
<tr>
<td>• Fully Qualified Domain Name (FQDN) is configured.</td>
<td></td>
</tr>
<tr>
<td>• UDS and EM services are running on all of the hub nodes in the ILS cluster.</td>
<td></td>
</tr>
<tr>
<td>• DNS primary and reverse resolution are working fine.</td>
<td></td>
</tr>
<tr>
<td>• Import consolidated Tomcat certificates from all the hub nodes.</td>
<td></td>
</tr>
<tr>
<td>Else, the &quot;version&quot; information will not get displayed in the Find and List Remote Clusters window even after rebooting the clusters or correcting the errors. The workaround is to remove the hub cluster from the ILS network, comply with the above requirements and add the hub cluster back into the ILS network.</td>
<td>ILS Network Capacities</td>
</tr>
<tr>
<td>ILS networking supports up to 10 hub clusters with 10 spoke clusters per hub, up to a total of 100 clusters per hub cluster. The design limits a maximum of 10 ILS clusters connections per hub cluster. A hub and spoke combination topology is used to avoid many TCP connections created within each cluster.</td>
<td></td>
</tr>
<tr>
<td>The specified number of hubs and spokes are only recommendations. Unified Communications Manager does not enforce to limit the numbers. However, to ensure optimum performance use this combination.</td>
<td></td>
</tr>
</tbody>
</table>

### ILS Troubleshooting

#### Local Cluster Cannot Connect to the ILS Network

To troubleshoot connection issues within the local cluster, open RTMT and run alarms and diagnostic traces on that publisher node.

If you receive an error message when trying to establish ILS between your clusters, you can try to restart the Cisco Intercluster Lookup service from Cisco Unified Serviceability Administration.

In addition, connection issues may arise if authentication is improperly configured between clusters. Check authentication in the following manner:

- If you are using TLS, make sure that all clusters in the network are using TLS and that Tomcat certificates have been exchanged for all the servers that need to communicate.

  **Note**

  Certificates exchanged using bulk certificate export, merge, and import can cause an untrusted ILS hub due to TLS errors.

- If you are using TCP password authentication, make sure that all ILS clusters are using TCP password authentication and that the same TCP password is assigned across the network.
Directory URIs Are Not Being Replicated Across the ILS Network

This error can occur for a variety of reasons. Check the following:

- Verify that all clusters in the network are configured to exchange global dial plan data. If a hub cluster is not configured to exchange global dial plan data, none of that hub’s spoke clusters will be able to exchange directory URI catalogs.

- Allow enough time for end-to-end replication based on synchronization intervals (set on the ILS Configuration page) that are configured for all the clusters involved in the path. All clusters in an ILS network are a maximum of three hops from every other cluster in the network.

- Use the `utils ils showpeerinfo` CLI command to monitor replication progress by looking at the USN values for the remote clusters.

- Increase speed of replication by changing the ILS Sync Throttle Service Parameter. Note that a low setting can affect system performance.

- Verify that all clusters in the ILS network have unique cluster IDs and that none of the clusters are configured with Stand Alone Cluster as its cluster ID. You can check Cluster IDs in Cisco Unified CM Administration under `System > Enterprise Parameters`.

Global Dial Plan Replication Is Configured, but Unified CM Still Cannot Place a Call to A Learned Directory URI or Learned Number in a Remote ILS Cluster

This condition can occur if ILS and Global Dial Plan Replication are enabled on all clusters in the network, but SIP route patterns that route to the route strings for the remote clusters have not been configured. Do the following:

- In the ILS Clusters and Global Dial Plan Imported Catalogs view in the ILS Configuration window, check the route string for the remote cluster.

- In the SIP Route Pattern configuration window, make sure that you have route patterns that map to the route strings for your remote clusters.

When advertising URI patterns (`user@domain`), in the SIP Profile Configuration window, make sure that the Dial String Interpretation field is set to Always treat all dial strings as URI addresses to prevent the devices to dial URI learned patterns with only numbers in the user section as Directory Number patterns. Alternatively, you can advertise only URI patterns with text strings in the user section through ILS.

ILS Global Dial Plan Replication Update is Cached Until Cisco Unified Communications Manager Database Replication is Not Corrected

The value of Last USN Data Received is Current and USN Data Synchronization Status is Up to date. However, the Learned URIs or Learned Patterns are not viewable on local cluster.

This condition occurs when database replication is not corrected. ILS caches the updates with learned URIs or patterns from remote clusters if database replication is not viewable on local cluster across all nodes. Unified Communications Manager can place a call to these learned URIs or patterns after the database replication is corrected.
Configure Global Dial Plan Replication

• Global Dial Plan Replication Overview, on page 183
• Global Dial Plan Replication Prerequisites, on page 185
• Global Dial Plan Replication Task Flow, on page 186

Global Dial Plan Replication Overview

Use global dial plan replication to create a global dial plan that spans across the Intercluster Lookup Service (ILS) network. When you enable Global Dial Plan Replication, you configure the dial plan component on one cluster, and ILS replicates that information throughout the ILS network.

When you enable Global Dial Plan Replication, each cluster in an ILS network advertises its global dial plan data, including the global dial plan data that was configured locally and any data that was learned from other clusters, to the ILS network. Global dial plan data includes the following:

• Directory universal resource indicators (URIs)
• Alternate numbers
• Advertised patterns
• PSTN failover
• Route strings
• Learned Global Dial Plan Data
• Imported Global Dial Plan Data

Directory URIs

ILS advertises the full catalog of locally configured directory URIs when you choose Advertise Globally via ILS option. See the URI Dialing Overview, on page 193 for more information on how to configure URI dialing.

Alternate Numbers

Alternate numbers allow you to configure globally routable numbers that can be dialed from anywhere within an ILS network. Cisco Unified Communications Manager allows you to create two types of alternate numbers:

• Enterprise alternate numbers
• +E.164 alternate numbers

**Advertised Patterns**

Advertised patterns allow you to create summarized routing instructions for a range of enterprise alternate numbers or +E.164 alternate numbers and replicate that pattern throughout an ILS network so that all clusters within the ILS network know the pattern. Advertised patterns prevent you from individually configuring routing information for each alternate number. Advertised patterns are never used by the local cluster on which they are configured; they are only used by remote clusters that learn the pattern through ILS. You can also configure Public Switched Telephone Network (PSTN) failover information for patterns that are advertised by ILS.

**PSTN Failovers**

Unified Communications Manager uses a PSTN failover number to reroute only those calls that are placed to patterns, alternate numbers, or directory URIs that were learned through ILS. Communications Manager does not reroute calls to the PSTN failover number for calls that are placed to locally configured patterns, alternate numbers, and directory URIs.

When you enable Global Dial Plan Replication, you can configure ILS to replicate a PSTN failover rule for learned directory URIs, learned numbers, and learned patterns. If the dial string for an outgoing call matches a learned pattern, learned alternate number, or learned directory URI, and Unified Communications Manager cannot route the call over a SIP trunk, Unified Communications Manager uses the calling party's Automatic Alternate Routing (AAR) CSS to reroute the call to the associated PSTN failover number.

**Route Strings**

ILS advertises the local route string to the ILS network. Each global dial plan data element associates to a route string that identifies the home cluster for that element. Remote clusters use the route string with a SIP route pattern to route to the various clusters in an ILS network. When a user in a remote cluster dials a directory URI or alternate number that was learned through ILS, Unified Communications Manager matches the associated route string to a SIP route pattern, and routes the call to the trunk that is specified by the SIP route pattern.

When a user assigns route string to a cluster, ILS associates that route string to all the global dial plan data that is local to that cluster (including locally configured directory URIs, alternate numbers, advertised patterns, and PSTN failover information).
If the SIP Route Pattern name contains dashes, you must ensure that there are no numerical digits between dashes. However, you can use a combination of letters and numbers or letters only, if there are more than one dash.

Examples of right and wrong SIP Route Patterns are listed in the following:

Correct Patterns:
- abc-1d-efg.xyz.com
- 123-abc-456.xyz.com

Incorrect Patterns:
- abc-123-def.xyz.com
- 1bc-2-3ef.xyz.com

**Learned Global Dial Plan Data**

Unified Communications Manager stores in the local database all global dial plan data that is learned through ILS. In addition to replicating locally configured data, ILS advertises all global dial plan data that the local cluster has learned from other clusters in the ILS network. This ensures that all advertised data reaches each cluster in the ILS network. Learned global dial plan data includes learned directory URIs, learned alternate numbers, learned patterns, learned PSTN failover rules, and learned route strings.

In Cisco Unified CM Administration, you can view the following types of learned global dial plan data:
- Learned Alternate Numbers
- Learned Enterprise and +E.164 Patterns
- Learned Directory URIs

**Imported Global Dial Plan Data**

Unified Communications Manager allows you to import global dial plan data from a CSV file into any hub cluster in an ILS network. ILS replicates the imported global dial plan data throughout the ILS network that allows you to interoperate Unified Communications Manager with a Cisco TelePresence Video Communications Server or a third-party call control system. Imported global dial plan data includes directory URIs, +E.164 patterns, and PSTN failover rules that were imported manually from a CSV file.

Imported data includes only global dial plan data that is imported manually into Unified Communications Manager. Imported global dial plan data does not include data that was learned through ILS.

**Global Dial Plan Replication Prerequisites**

Follow the procedures to set up an ILS network in the ILS Configuration Task Flow, on page 172.
# Global Dial Plan Replication Task Flow

## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Enable ILS Support for Global Dial Plan Replication, on page 187.</td>
<td>Enable ILS support for Global Dial Plan Replication to share dial plan information between participating ILS-enabled clusters.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Set Up Alternate Number, on page 187.</td>
<td>(Optional) If you want to set up alternate numbers that you can dial between clusters, configure alternate number replication.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Set Up Advertised Pattern for Alternate Numbers, on page 188.</td>
<td>(Optional) If you want to summarize your alternate numbers with a pattern, set up an advertised pattern, and assign a PSTN failover rule for the pattern.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Set Up PSTN Failover, on page 189.</td>
<td>(Optional) If you want to set up a PSTN failover number for specific directory URIs or alternate numbers, assign an alternate number as the PSTN failover number for all the directory URIs and alternate numbers that are associated to a specific directory number.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Assign Partitions for Learned Numbers and Patterns, on page 189.</td>
<td>(Optional) Assign route partitions to the alternate numbers and patterns that the local cluster learns through ILS.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Block a Learned Pattern, on page 190.</td>
<td>(Optional) To prevent a local Unified Communications Manager cluster from routing calls to a learned alternate number or learned alternate number pattern, you can configure a local blocking rule on that cluster.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Set Database Limits for Learned Data, on page 191.</td>
<td>Set a database limit to determine the number of learned objects that Unified Communications Manager can write to the local database.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Import Global Dial Plan Data, on page 191.</td>
<td>(Optional) If you want your ILS network to interoperate with a Cisco TelePresence Video Communication Server or third-party call control system, import directory URI catalogs from a CSV file for the other system into any hub cluster in the ILS network.</td>
</tr>
</tbody>
</table>

## What to do next

If you want to dial directory universal resource indicators (URIs) across clusters, set up URI dialing in the local cluster. For details, see the URI Dialing Overview, on page 193.
Enable ILS Support for Global Dial Plan Replication

To enable ILS support for Global Dial Plan Replication in the local cluster, follow this procedure:

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Log in to the Cisco Unified Communications Manager publisher node.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From Cisco Unified CM Administration, choose Advanced Features &gt; ILS Configuration.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Check the Exchange Global Dial Plan Replication Data with Remote Clusters check box.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Advertised Route String text box, enter a route string for the local cluster.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Set Up Alternate Number

Create an enterprise alternate number or +E.164 alternate number and associate the alternate number with a directory number. When you dial the alternate number, the phone that is registered to the associated directory number, rings.

**Note**

Each alternate number that you set up must associate with a single directory number. However, that directory number can associate to both an enterprise alternate number and a +E.164 alternate number at the same time.

**Before you begin**

Enable ILS Support for Global Dial Plan Replication, on page 187.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose Call Routing &gt; Directory Number.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Find and List Directory Numbers window, find and select the directory number to which you want to associate the alternate number.</td>
</tr>
</tbody>
</table>
| Step 3 | From the Directory Number Configuration window, click one of the following options depending on the type of alternate number that you want to assign:  
  - Add Enterprise Alternate Number.  
  - Add +E.164 Alternate Number. |
| Step 4 | In the Number Mask field, enter the number mask that you want to apply to the directory number.  
The Alternate Number field displays how the alternate number appears after Cisco Unified Communications Manager applies the number mask. |
| Step 5 | (Optional) If you want to enable local routing for the alternate number, perform the following steps:  
a) Check the Add to Local Route Partition check box. |
b) From the **Route Partition** drop-down list, choose a route partition that is assigned to a local calling search space.

**Step 6**  
(Optional) If you want to use a number pattern to set up intercluster routing for this alternate number, click **Save**.

**Step 7**  
(Optional) If you want to set up intercluster routing for this alternate number, check the **Advertise Globally via ILS** check box for this alternate number.

**Step 8**  
(Optional) If you want to assign a PSTN failover number to this alternate number, from the **PSTN failover** drop-down list, assign a number as the PSTN failover.

**Step 9**  
Click **Save**.

---

**What to do next**

Set Up Advertised Pattern for Alternate Numbers, on page 188.

---

**Set Up Advertised Pattern for Alternate Numbers**

Use advertised patterns to summarize a range of Enterprise alternate numbers or E.164 alternate numbers. You can advertise the pattern to the ILS network to enable intercluster calling to numbers that match the pattern.

**Procedure**

---

**Step 1**  
From Cisco Unified CM Administration, choose **Call Routing > Global Dial Plan Replication > Advertised Patterns**.

**Step 2**  
From the **Find and List Advertised Patterns** window, do either of the following:

- Click **Find** and select an existing pattern.
- Click **Add New** to create a new pattern.

**Step 3**  
In the **Pattern** field, enter the number pattern. For example, 54XXX summarizes a range of numbers between 54000 - 54999.

**Step 4**  
In the **Pattern Type** field, select the pattern type: **Enterprise Number Pattern** or **E.164 Number Pattern**.

**Step 5**  
From the radio buttons, select whether you want to apply a PSTN Failover.

- **Don't use PSTN Failover**
- **Use Pattern as PSTN Failover**
- **Apply Strip Digits and Prepend Digits to Pattern and Use for PSTN Failover**—If you choose this option, enter the digits in the **PSTN Failover Strip Digits** and **PSTN Failover Prepend Digits** fields.

**Step 6**  
Click **Save**.
Set Up PSTN Failover

Perform the following procedure to assign a PSTN failover number for directory URIs or alternate numbers and advertise that PSTN failover number to the ILS network. Remote clusters can use the PSTN failover number for calls to learned directory URIs or learned alternate numbers.

Before you begin

Set Up Advertised Pattern for Alternate Numbers, on page 188.

Procedure

Step 1

From Cisco Unified CM Administration, choose Call Routing > Directory Number.

Step 2

From the Find and List Directory Numbers window, find and select the directory number that is associated to the directory URI or alternate number for which you want to assign a PSTN failover number. The appears.

Step 3

(Optional) If the alternate number that you want to use as the PSTN failover does not exist, in the Directory Number Configuration window, choose one of the following options depending on the type of alternate number that you want to assign:

- Add Enterprise Alternate Number.
- Add +E.164 Alternate Number.

Step 4

In the PSTN Failover drop-down list, choose the alternate number that you want to use as the PSTN failover. Click Save.

Step 5

Cisco Unified Communications Manager associates that PSTN failover number to that directory number. Global Dial Plan Replication advertises that number to the ILS network as the PSTN failover number for all the directory URIs and alternate numbers that are associated to that directory number.

What to do next

Assign Partitions for Learned Numbers and Patterns, on page 189.

Assign Partitions for Learned Numbers and Patterns

You must assign learned numbers and learned patterns to a partition. You can define your own partitions or use the predefined default partitions. Unified Communications Manager is installed with the following predefined partitions for learned alternate numbers and number patterns:

- Global Learned Enterprise Numbers.
- Global Learned E.164 Numbers.
- Global Learned Enterprise Patterns.
- Global Learned E.164 Patterns.
You cannot assign a learned number or learned pattern to a NULL partition.

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose Call Routing > Global Dial Plan Replication > Partitions for Learned Numbers and Patterns.

**Step 2** Configure the fields in the Partitions for Learned Numbers and Patterns window. For more information on the fields and their configuration options, see the system Online Help.

**Step 3**  Click Save.

Note The route partition must also exist in the calling search space that is used by the calling party in order for calls to be placed to numbers in the partition.

---

**Block a Learned Pattern**

Complete this optional task if you want to set up a blocking rule that prevents the local cluster from routing calls to specific enterprise alternate numbers, +E.164 alternate numbers, or number patterns that were learned through the ILS.

Before routing a call to a learned number or learned pattern, ILS checks to see if a local blocking rule matches the dial string. If the blocking rule matches, Unified Communications Manager does not route the call.

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose Call Routing > Global Dial Plan Replication > Block Learned Numbers and Patterns.

**Step 2**  Perform one of the following tasks:

- Click Find and select an existing blocking rule to edit.
- Click Add New to create a new blocking rule.

**Step 3**  In the Pattern field, enter the pattern or number that you want to block. For example, 206XXXXXXX can be used to block calls to 2065551212.

**Step 4**  If you want to block calls based on the dial string prefix, enter the Prefix.

**Step 5**  If you want to block calls from being sent to a specific cluster, enter the Cluster ID of the cluster.

**Step 6**  From the Pattern Type drop-down list, select how you want to apply the blocking rule:

- **Any**—Choose this option if the blocking rule applies to both enterprise number patterns and +E.164 patterns.
- **Enterprise Pattern**—Choose this option if the blocking rule applies to enterprise number patterns only.
- **+E.164 Pattern**—Choose this option if the blocking rule applies to +E.164 number patterns only.
Set Database Limits for Learned Data

Set a database limit to determine the number of learned objects that Unified Communications Manager can write to the local database.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **System > Service Parameters**.

**Step 2** Choose the **Server** where you want to configure the parameter.

**Step 3** From the **Service** drop-down list, choose **Cisco Intercluster Lookup Service (Active)**. If the service does not appear as active, ensure that the service is activated in Cisco Unified Serviceability.

**Step 4** Under **Clusterwide Parameters (ILS)** section, set an upper limit for the **ILS Max Number of Learned Objects in Database** service parameter.

**Step 5** Click **Save**.

**Note**

This service parameter determines the maximum number of entries that Unified Communications Manager can write to the database for data that is learned through ILS. The default value of the service parameter is 100,000 while the maximum value of the service parameter is 1,000,000.

If you reduce the service parameter to a value that is lower than the current number of ILS-learned entries that are saved in the database, Unified Communications Manager does not write additional ILS learned objects to the database. However, the existing database entries remain.

Import Global Dial Plan Data

Use this procedure if you are interoperating with a Cisco TelePresence Video Communications Server, a third-party call control system, or another system that is not running ILS. You can import a catalog of directory URIs, +E.164 patterns and PSTN failover rules from the other system into a hub cluster in the ILS network. ILS replicates the catalog throughout the ILS network so that the clusters can place calls to the other system.

**Before you begin**

Export your dial plan catalogs from the other system to a CSV file.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Call Routing > Global Dial Plan Replication > Imported Global Dial Plan Catalog**.

**Step 2** From the **Find and List Imported Global Dial Plan Catalogs** window, perform one of the following tasks:
• Click **Find** and select an existing catalog from the resulting list.
• Click **Add New** to add a new catalog.

**Step 3**
From the **Imported Global Dial Plan Catalog Settings** window, in the **Name** field, enter a unique name to identify the catalog that you want to import.

**Step 4**
(Optional) In the **Description** field, enter a description of the catalog.

**Step 5**
In the **Route String** field, create a route string for the system from which you are importing the catalog.

**Note** Route strings can be up to 250 alphanumeric characters long and can include dots and dashes.

**Step 6**
Click **Save**.

**Step 7**
From Cisco Unified CM Administration, choose **Bulk Administration** > **Upload/Download Files**.
• Click **Add New**.
• Click **Browse** and select the CSV file for the catalog that you want to import.

**Note** Ensure that the CSV file that you use for the import is compatible with the version of Cisco Unified Communications Manager. For example, a CSV file that is compatible to import into Version 9.0(1) is not compatible with Version 10.0(1).

**Step 8**
In the **Select the Target** drop-down list, select **Imported Directory URIs and Patterns**.

**Step 9**
In the **Select Transaction Type** drop-down list, select **Insert Imported Directory URIs and Patterns**.

**Step 10**
Click **Save**.

**Step 11**
From Cisco Unified CM Administration, choose **Bulk Administration** > **Directory URIs and Patterns** > **Insert Imported Directory URIs and Patterns**.

**Step 12**
In the **File Name** drop-down list, choose the CSV file that contains the catalog that you want to import.

**Step 13**
In the **Imported Directory URI Catalog** drop-down list, choose the catalog that you named in the **Imported Global Dial Plan Catalog** window.

**Step 14**
In the **Job Description** text box, enter a name for the job that you are about to run.

**Step 15**
Perform one of the following steps:
• If you want to run the job now, select the **Run Immediately** option, and click **Submit**.
• If you want to schedule the job to run at a specified time, select the **Run Later** radio button and click **Submit**.

**Note** If you choose the **Run Later** option, you must use the Bulk Administration Job Scheduler to schedule when the job runs.

Cisco Unified Communications Manager saves all imported +E.164 patterns to the Global Learned +E.164 Patterns partition.

---

**Note** You can also export all locally configured directory URIs, +E.164 number patterns, and their associated PSTN failover rules to a CSV file that you can import into the other call control system. Refer to the menus at **Bulk Administration** > **Directory URIs and Patterns** > **Export Local Directory URIs and Patterns** for details.
CHAPTER 27

Configure URI Dialing

- URI Dialing Overview, on page 193
- URI Dialing Prerequisites, on page 194
- URI Dialing Configuration Task Flow, on page 195

URI Dialing Overview

Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. A directory URI is a uniform resource identifier, a string of characters that can be used to identify a directory number. Directory URIs look like email addresses and follow the username@host format where the host portion is an IPv4 address or a fully qualified domain name. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support directory URIs.

Directory URI Format

Directory URIs are alphanumeric strings that consist of a user and a host address separated by the @ symbol. Cisco Unified Communications Manager supports the following formats for directory URIs:

- user@domain (for example, joe@cisco.com)
- user@ip_address (for example, joe@10.10.10.1)

The system supports the following formats in the user portion of a directory URI (the portion before the @ symbol):

- Accepted characters are a-z, A-Z, 0-9, !, $, %, &, *, +, -, =, \, ?, ', ,, ., /.
- The user portion has a maximum length of 47 characters.
- Cisco Unified Communications Manager automatically applies percent encoding to the following characters when the directory URI is saved in the database:
  
  # % ^ ` { } | : ” < > [ ] \ ^ and spaces.
The user portion of a directory URI is case sensitive by default. You can edit the user portion to be case insensitive by editing the **URI Lookup Policy** enterprise parameter.

When you apply percent encoding, the digit length of the directory URI increases. For example, if you input joe smith#@cisco.com (20 characters) as a directory URI, Cisco Unified Communications Manager stores the directory URI in the database as joe%20smith%23@cisco.com (24 characters). Due to database restrictions, the **Directory URI** field has a maximum length of 254 characters.

Cisco Unified Communications Manager supports the following formats in the host portion of a directory URI (the portion after the @ symbol):

- Supports IPv4 addresses or fully qualified domain names.
- Accepted characters are alphanumeric characters, hyphens (-), and dots (.).
- The host portion cannot start or end with a hyphen (-).
- The host portion cannot have two dots in a row.
- The host portion has a minimum length of two characters.
- The host portion is not case sensitive.

Within **Cisco Unified Communications Manager Administration**, when you use Bulk Administration to import a CSV file that contains directory URIs with embedded double quotes and commas, you must enclose the entire directory URI in double quotes (".

**URI Dialing Prerequisites**

Before you configure the URI dialing, you must set up an ILS network and enable Global Dial Plan Replication in the ILS network. Refer the following sections to complete this task:

- Global Dial Plan Replication Task Flow, on page 186
- ILS Configuration Task Flow, on page 172
## URI Dialing Configuration Task Flow

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 1 | Assign directory URIs to the local cluster in the network:  
- Assign Directory URI to Users, on page 196  
- Associate Directory URI with Directory Numbers, on page 196 | Provision end users into your system and assign directory URI to those end users. Also, configure a directory number and associate a directory URI with that directory number.  
**Note** For both end user configuration and directory number configuration, you can also use Bulk Administration to import end users, directory URIs, directory numbers, and phones into Cisco Unified Communications Manager. For more information, see the **Bulk Administration Guide for Cisco Unified Communications Manager** at [http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html](http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html). |
| Step 2 | Assign Default Directory URI Partition, on page 197 | Assign the default directory URI partition to an existing partition that is located in a calling search space. |
| Step 3 | Configure SIP Profiles for URI Dialing, on page 197 | Configure the SIP profiles to configure intercluster URI dialing in your network. |
| Step 4 | Configure SIP Trunks for URI Dialing, on page 198 | Configure whether Cisco Unified Communications Manager inserts a directory number, a directory URI, or a blended address for outgoing SIP messages. |
| Step 5 | Configure SIP Route Patterns, on page 199 | Configure SIP route patterns to route intercluster directory URI calls. |
| Step 6 | Repeat step 1 to step 5 for all the clusters in your ILS network. | Perform this step if you have multiple clusters in your ILS network. |
| Step 7 | Import Directory URI Catalogs, on page 199 | (Optional) If you want to place directory URI calls to a Cisco TelePresence Video Communications Server, or a third-party call control system, import directory URI catalogs from a CSV file for the other system into any hub cluster in the ILS network. |
Assign Directory URI to Users

Perform the following steps to assign a directory URI to an end user.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose User Management &gt; End User.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the Find and List Users window, specify the search criteria and click Find.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose a user from the resulting list. The End User Configuration window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Directory URI field, enter a directory URI that you want to associate to this end user. A directory URI looks like an email address and follows the user@host format.</td>
</tr>
</tbody>
</table>

**Note** If you enter a directory URI and also enter a directory number in the Primary Extension field, this directory URI automatically becomes the primary directory URI that is associated to that directory number.

| Step 5 | Click Save. |

What to do next

Associate Directory URI with Directory Numbers, on page 196

Related Topics

Directory URI Format, on page 193

Associate Directory URI with Directory Numbers

Perform the following procedure to associate a directory URI with a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager allows you to dial that phone using the directory URI.

Before you begin

Assign Directory URI to Users, on page 196

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose Device &gt; Phone. The Find and List Phones window appears.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Specify the filter criteria and click Find.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click on the device for which you want to associate the directory number. The Phone Configuration window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Association pane:</td>
</tr>
<tr>
<td></td>
<td>• Click on an existing directory number.</td>
</tr>
<tr>
<td></td>
<td>• Click on Add a new DN if no directory numbers are configured.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the Directory Number Configuration window, enter the directory URI address in the URI text box.</td>
</tr>
</tbody>
</table>
Step 6  From the Partition drop-down list, choose the partition to which the directory URI belongs. Ensure that the directory URI that you enter is unique within the partition that you choose. If you do not want to restrict access to the URI, choose None for the partition.

Step 7  Click Save.

What to do next
Assign Default Directory URI Partition, on page 197

Assign Default Directory URI Partition

Perform the following procedure to assign a default directory URI partition.

Before you begin
Assign Default Directory URI Partition, on page 197

Procedure

Step 1  In Cisco Unified CM Administration, choose System > Enterprise Parameters. The Enterprise Parameters Configuration window appears.

Step 2  For the Directory URI Alias Partition in the End User Parameters area, choose an existing partition that is in an existing calling search space.

Step 3  Click Save.

What to do next
Configure SIP Profiles for URI Dialing, on page 197

Configure SIP Profiles for URI Dialing

Before you begin
Assign Default Directory URI Partition, on page 197

Procedure

Step 1  In Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile. The Find and List SIP Profiles window appears.

Step 2  Enter the appropriate search criteria and click Find. A list of existing SIP profiles appear.

Step 3  Select the SIP profile that you want to view. The SIP Profile Configuration window appears.

Step 4  From the Dial String Interpretation drop-down list, choose one of the following options:
• **Always treat all dial strings as URI addresses**—Select this option to treat the address of incoming calls as URI addresses.

• **Phone number consists of characters 0–9, A–D, *, and + (others treated as URI addresses)**—Select this option to treat the incoming call as a directory number if all the characters in the user portion of the SIP identity header fall within this range. If the user portion of the address uses any characters that do not fall within this range, the address is treated as a URI.

• **Phone number consists of characters 0-9, *, and + (others treated as URI addresses)**—Select this option to treat the incoming call as a directory number if all the characters in the user portion of the SIP identity header fall within this range. If the user portion of the address uses any characters that do not fall within this range, the address is treated as a URI.

**Step 5** Check the Use Fully Qualified Domain Name in SIP Requests check box for all the SIP profiles in your network.

**Step 6** Click Apply Config.

---

**Configure SIP Trunks for URI Dialing**

If you are deploying URI dialing, configure the contact header addressing policy for the SIP trunks in your network. Cisco Unified Communications Manager can insert a directory number, directory URI, or a blended address that includes both the directory number and directory URI in the SIP identity headers for outgoing SIP messages.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Trunk**.

**Step 2** Click **Find** and select an existing SIP trunk.

**Step 3** In the **Outbound Calls** area, select one of the following from the Calling and Connected Party Info Format drop-down list:

• **Deliver DN only in connected party**—In outgoing SIP messages, Unified Communications Manager inserts the calling party’s directory number in the SIP contact header information. This is the default setting.

• **Deliver URI only in connected party, if available**—In outgoing SIP messages, Unified Communications Manager inserts the sending party’s directory URI in the SIP contact header. If a directory URI is not available, Unified Communications Manager inserts the directory number instead.

• **Deliver URI and DN in connected party, if available**—In outgoing SIP messages, Unified Communications Manager inserts a blended address that includes the calling party’s directory URI and directory number in the SIP contact headers. If a directory URI is not available, Unified Communications Manager includes the directory number only.

**Step 4** Click **Save**.
Configure SIP Route Patterns

You must configure SIP route patterns to route intercluster directory URI calls.

Follow these steps to configure SIP route patterns.

Before you begin

Configure SIP Trunks for URI Dialing, on page 198

Procedure

Step 1 In Cisco Unified CM Administration, choose Call Routing > SIP Route Pattern.
Step 2 Choose one of the following options:
   • To add a new SIP route pattern, click the Add New button.
   • To modify the settings for an existing SIP route pattern, enter the search criteria, click Find, and choose a SIP route pattern from the resulting list.
Step 3 Configure the fields in the SIP Route Pattern Configuration window. See the online help for more information about the fields and their configuration options.
Step 4 Click Save.

What to do next

(Optional) Import Directory URI Catalogs, on page 199

Import Directory URI Catalogs

Cisco Unified Communications Manager allows you to import global dial plan data from a CSV file into any hub cluster in an ILS network and ILS replicates the imported global dial plan data throughout the ILS network allowing you to interoperate Cisco Unified Communications Manager with a Cisco TelePresence Video Communications Server or a third-party call control system.

(Optional) To import directory URI catalogs, follow this procedure:

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Global Dial Plan Replication > Imported Global Dial Plan Catalog.
Step 2 From the Find and List Imported Global Dial Plan Catalogs window, perform one of the following tasks:
   • Click Find and select an existing catalog from the resulting list.
   • Click Add New to add a new catalog.
Step 3 From the Imported Global Dial Plan Catalog Settings window, in the Name field, enter a unique name to identify the catalog that you want to import.
Step 4 (Optional) In the Description field, enter a description of the catalog.
Step 5  In the Route String field, create a route string for the system from which you are importing the catalog.

Note  Route strings can be up to 250 alphanumeric characters long and can include dots and dashes.

Step 6  Click Save.

Step 7  From Cisco Unified CM Administration, choose Bulk Administration > Upload/Download Files.

  • Click Add New.
  • Click Browse and select the CSV file for the catalog that you want to import.

Note  Ensure that the CSV file that you use for the import is compatible with the version of Cisco Unified Communications Manager. For example, a CSV file that is compatible to import into Version 9.0(1) is not compatible with Version 10.0(1).

Step 8  In the Select the Target drop-down list, select Imported Directory URIs and Patterns.

Step 9  In the Select Transaction Type drop-down list, select Insert Imported Directory URIs and Patterns.

Step 10  Click Save.

Step 11  From Cisco Unified CM Administration, choose Bulk Administration > Directory URIs and Patterns > Insert Imported Directory URIs and Patterns.

Step 12  In the File Name drop-down list, choose the CSV file that contains the catalog that you want to import.

Step 13  In the Imported Directory URI Catalog drop-down list, choose the catalog that you named in the Imported Global Dial Plan Catalog window.

Step 14  In the Job Description text box, enter a name for the job that you are about to run.

Step 15  Perform one of the following steps:

  • If you want to run the job now, select the Run Immediately option, and click Submit.
  • If you want to schedule the job to run at a specified time, select the Run Later radio button and click Submit.

Note  If you choose the Run Later option, you must use the Bulk Administration Job Scheduler to schedule when the job runs.

Cisco Unified Communications Manager saves all imported +E.164 patterns to the Global Learned +E.164 Patterns partition.
PART IV

Configure Call Admission Control

- Call Admission Control Overview, on page 203
- Configure Enhanced Locations Call Admission Control, on page 205
- Configure Resource Reservation Protocol, on page 213
About Call Admission Control

Use call admission control (CAC) to regulate voice quality over a WAN link.

Voice quality can degrade when too many active calls exist on a link and the amount of bandwidth is oversubscribed. Call admission control regulates voice quality by limiting the number of calls that can be active at the same time on a particular link. Call admission control does not guarantee a particular level of audio quality on the link, but it does allow you to regulate the amount of bandwidth that active calls on the link consume.

Call admission control operates by rejecting a call for bandwidth and policy reasons. When a call is rejected due to call admission control, the phone of the called party does not ring, and the caller receives a busy tone. The caller also receives a message on their phone, such as “Not enough bandwidth.” If you have enabled automated alternate routing (AAR), call admission control automatically diverts calls to alternate public switched telephone network (PSTN) routes when WAN bandwidth is not available.

Call Admission Control Configuration

Choose from one of the following task flows to implement call admission control (CAC).

<table>
<thead>
<tr>
<th>Task Flow</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enhanced Locations Call Admission Control Task Flow</td>
<td>Use enhanced locations CAC in distributed deployments, where multiple clusters manage devices in the same physical sites using the same WAN uplinks. Enhanced locations CAC lets you regulate voice quality by limiting the amount of bandwidth that is available for calls over links between the locations. It also allows you to control call admissions for immersive video calls, such as TelePresence, separately from other video calls.</td>
</tr>
<tr>
<td>RSVP Configuration Task Flow</td>
<td>Use RSVP to implement call admission control in complex, multi-tiered topologies that include IP telephony and videoconferencing applications. RSVP is also able to handle dynamic changes to bandwidth.</td>
</tr>
</tbody>
</table>
CHAPTER 29

Configure Enhanced Locations Call Admission Control

- Enhanced Locations Call Admission Control Overview, on page 205
- Enhanced Locations Call Admission Control Prerequisites, on page 206
- Enhanced Locations Call Admission Control Task Flow, on page 207
- Enhanced Locations Call Admission Control Interactions and Restrictions, on page 212

Enhanced Locations Call Admission Control Overview

Enhanced locations call admission control (CAC) provides control over WAN bandwidth in complex WAN topologies as well as distributed deployments, where multiple clusters manage devices in the same physical sites using the same uplinks. Enhanced locations CAC also allows you to control call admissions for immersive video calls, such as TelePresence separately from other video calls.

You can effectively share locations between clusters by enabling the clusters to communicate with one another to reserve, release, and adjust allocated bandwidth for the same locations across clusters.

Network Modeling

To define how your system handles media, you structure your network model around the concepts of locations and links.

A location represents a local area network (LAN). It could contain endpoints or simply serve as a transit location between links for wide area network (WAN) network modeling.

Links interconnect locations and are used to define bandwidth available between locations. Links represent the WAN link.

Weights are measurements of bandwidth pathways. These are used on links to provide a cost to the effective path. Weights are provided when there is more than one path between any two locations.

Your system calculates shortest paths (least cost) from all locations to all locations and builds effective paths. These have the least overall weight and are the most efficient pathways.

Locations Media Resource Audio Bit Rate Policy service parameter determines the bit rate value to deduct from the audio bandwidth pools within and between the locations of the parties for an audio-only call when a media resource such as a transcoder is inserted into the media path and for more complex scenarios. This service parameter does not have any impact if there is no media in one of the call legs. In such cases, location bandwidth manager deducts the maximum hop bandwidth that is configured for the source destination from the available bandwidth of that location.
Location Bandwidth Manager

The location bandwidth manager (LBM) service computes the effective path from source location to destination location. It provides useful functions behind the scenes, such as handling bandwidth requests from Unified Communications Manager call control and replicating bandwidth information within the cluster and between clusters. You can find the configured and realtime information that this function provides in Serviceability Administration.

Locations Media Resource Audio Bit Rate Policy service parameter determines the bit rate value to deduct from the audio bandwidth pools within and between the locations of the parties for an audio-only call when a Media Resource such as a transcoder is inserted into the media path and for more complex scenarios. This service parameter does not have any impact if there is no media in one of the call legs. In such cases, location bandwidth manager deducts the maximum hop bandwidth configured for the source destination from the available bandwidth of that location.

Intercluster Enhanced Locations Call Admission Control

The intercluster function extends enhanced locations CAC network modeling across multiple clusters. Each cluster manages its own network topology. They then propagate their topologies to other clusters that are configured in the LBM intercluster replication network.

A shared location is a location that is configured with the same name on clusters participating in a LBM replication network.

This type of location serves the following purposes:

- Enables clusters to share their respective configured topologies with one another
- Lets multiple clusters perform CAC on the same locations

Enhanced Locations Call Admission Control Prerequisites

- Unified Communications Manager and location bandwidth manager (LBM) manage bandwidth for all types of devices, including IP phones, gateways, and H.323 and SIP trunk destinations. However, intercluster enhanced locations CAC requires SIP intercluster trunks that are assigned to the system shadow location, which is a special location that has no links to other locations and no bandwidth allocations. All other types of devices are supported only when assigned to ordinary (fixed) locations.

- Unified Communications Manager and LBM do not manage bandwidth for media resources. For cases in which media resources change the bandwidth requirement for a call, you can change a global parameter setting that determines whether the minimum or maximum bandwidth is reserved.
## Enhanced Locations Call Admission Control Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Activate the LBM Service, on page 208</td>
<td>Verify whether the Cisco Location Bandwidth Manager service is activated. For a new system install, you must manually enable the service on the desired nodes. For enhanced locations CAC to work properly, one instance of this service must run on each cluster.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Create an LBM Group, on page 208</td>
<td>If LBM is not running on the same node, configure an LBM group and assign the LBM group to the server. The LBM group lets you optimize network delay and performance. Each server must communicate with an LBM service to determine the available bandwidth for each call and to deduct bandwidth for the duration of each call.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Locations and Location Links, on page 209</td>
<td>Configure locations to implement call admission control in a centralized call-processing system. A location represents a local area network (LAN), and can contain endpoints or simply serve as a transit location between links for wide area network (WAN) network modeling. Locations provide bandwidth accounting within a location as well as in or out of a location. Links provide bandwidth accounting between locations and interconnect locations.</td>
</tr>
<tr>
<td>Step 4</td>
<td>(Optional) Assign Intra-Location Bandwidth, on page 209</td>
<td>Assign intra-location bandwidth to the location, if you do not want to use the default of unlimited bandwidth. By default, when you create a new location, a link from the newly added location to the Hub_None is added as well, with unlimited audio bandwidth, 384 kbps video bandwidth and 384 kbps immersive video bandwidth. You can adjust this allotment to match your network model.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Establish External Communication, on page 210</td>
<td>Configure the LBM hub group to allow the LBM servers acting as hubs to find LBM servers in remote clusters. This step establishes external communication with those clusters. An LBM service becomes a hub when an LBM hub group is assigned to it. Any LBM servers that are assigned an LBM hub group establish...</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>----------------------------------------------------------------------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
<td></td>
</tr>
<tr>
<td>Step 6 Configure the SIP Intercluster Trunk for Enhanced Location Call Admission Control, on page 210</td>
<td>Assign a SIP intercluster trunk (ICT) to the shadow location to establish proper intercluster operation. SIP trunks that are linked to devices with a specific location, such as SIP gateways, can be assigned to ordinary locations. A shadow location is a special location that contains no links to other locations and no bandwidth allocations.</td>
<td></td>
</tr>
<tr>
<td>Step 7 (Optional) Deduct Audio Bandwidth from Audio Pool for Video Calls, on page 211</td>
<td>Use this procedure if you want to split the audio and video bandwidth deductions into separate pools for video calls. By default, the system deducts the bandwidth requirement for both the audio stream and video stream from the video pool for video calls.</td>
<td></td>
</tr>
</tbody>
</table>

**Activate the LBM Service**

Verify whether the Cisco Location Bandwidth Manager service is activated. For a new system install, you must manually enable the service on the desired nodes. For enhanced locations CAC to work properly, one instance of this service must run on each cluster.

**Procedure**

- **Step 1** From Cisco Unified Serviceability, choose **Tools > Service Activation**.
- **Step 2** From the **Server** drop-down list, choose a server, and then click **Go**.
- **Step 3** If needed, check the **Cisco Location Bandwidth Manager** check box.
- **Step 4** Click **Save**.

**Create an LBM Group**

If LBM is not running on the same node, configure an LBM group and assign the LBM group to the server. The LBM group lets you optimize network delay and performance. Each server must communicate with an LBM service to determine the available bandwidth for each call and to deduct bandwidth for the duration of each call.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose **System > Location Info > Location Bandwidth Manager Group**.
Configure Locations and Location Links

Configure locations to implement call admission control in a centralized call-processing system. A location represents a local area network (LAN), and can contain endpoints or simply serve as a transit location between links for wide area network (WAN) network modeling. Locations provide bandwidth accounting within a location as well as in or out of a location. Links provide bandwidth accounting between locations and interconnect locations.

Procedure

1. From Cisco Unified CM Administration, choose System > Location Info > Location.
2. Perform one of the following tasks:
   - Click Find and then choose an existing location from the resulting list, to modify the settings for an existing location.
   - Click Add New to add a new location.
3. Configure the fields on the Location Configuration window. For more information on the fields and their configuration options, see the system Online Help.
4. Click Save.

Assign Intra-Location Bandwidth

Assign intra-location bandwidth to the location, if you do not want to use the default of unlimited bandwidth. By default, when you create a new location, a link from the newly added location to the Hub_None is added as well, with unlimited audio bandwidth, 384 kbps video bandwidth and 384 kbps immersive video bandwidth. You can adjust this allotment to match your network model.

Tip

If the audio quality is poor or choppy, lower the bandwidth setting. For example, for ISDN use multiples of 56 kbps or 64 kbps.
## Establish External Communication

Configure the LBM hub group to allow the LBM servers acting as hubs to find LBM servers in remote clusters. This step establishes external communication with those clusters. An LBM service becomes a hub when an LBM hub group is assigned to it. Any LBM servers that are assigned an LBM hub group establish communication with all other LBM servers that are assigned the same or an overlapping LBM hub group.

### Procedure

**Step 1** From Cisco Unified CM Administration, choose **System > Location Info > Location Bandwidth Manager (LBM) Intercluster Replication Group**.

**Step 2** Perform one of the following tasks:

- Click **Find** to modify the settings for an LBM intercluster replication group, and choose an existing LBM intercluster replication group from the resulting list.
- Click **Add New** to add a new LBM intercluster replication group.

**Step 3** Configure the fields on the **Location Bandwidth Manager Intercluster Replication Group Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 4** Click **Save**.

## Configure the SIP Intercluster Trunk for Enhanced Location Call Admission Control

Assign a SIP intercluster trunk (ICT) to the shadow location to establish proper intercluster operation. SIP trunks that are linked to devices with a specific location, such as SIP gateways, can be assigned to ordinary locations. A shadow location is a special location that contains no links to other locations and no bandwidth allocations.

### Procedure

**Step 1** From Cisco Unified CM Administration, choose **System > Location Info > Location Bandwidth Manager (LBM) Intercluster Replication Group**.

**Step 2** Perform one of the following tasks:

- Click **Find** to modify the settings for an LBM intercluster replication group, and choose an existing LBM intercluster replication group from the resulting list.
- Click **Add New** to add a new LBM intercluster replication group.

**Step 3** Configure the fields on the **Location Bandwidth Manager Intercluster Replication Group Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 4** Click **Save**.
Before you begin

- A configured SIP intercluster trunk. See SIP Trunk Configuration Task Flow, on page 97 for more information.

- Establish External Communication, on page 210

Procedure

**Step 1**  From Cisco Unified CM Administration, choose **Device > Trunk**.

**Step 2**  Enter search criteria, click **Find**, and then choose an existing SIP intercluster trunk from the resulting list.

**Step 3**  From the **Location** drop-down list, choose **Shadow**.

**Step 4**  Click **Save**.

---

**Deduct Audio Bandwidth from Audio Pool for Video Calls**

Use this procedure if you want to split the audio and video bandwidth deductions into separate pools for video calls. By default, the system deducts the bandwidth requirement for both the audio stream and video stream from the video pool for video calls.

**Note**  When you enable this feature, CAC includes the bandwidth required for the IP/UDP network overhead in the audio bandwidth deduction. This audio bandwidth deduction equates to the audio bit rate plus the IP/UDP network overhead bandwidth requirement. The video bandwidth deduction is the video bit rate only.

Procedure

**Step 1**  From Cisco Unified CM Administration, choose **System > Service Parameters**.

**Step 2**  From the **Server** drop-down list, choose the publisher node.

**Step 3**  From the **Service** drop-down list, choose **Cisco CallManager**.

**Step 4**  From the **Clusterwide Parameters (Call Admission Control)** area, set the value of the **Deduct Audio Bandwidth Portion from Audio Pool for a Video Call** service parameter to **True**.

**Step 5**  Click **Save**.
Enhanced Locations Call Admission Control Interactions and Restrictions

Enhanced Locations Call Admission Control Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bandwidth</td>
<td>If there is a conflict in bandwidth capacity or weight assignment on the common links or locations, the local cluster uses the minimum of the assigned values.</td>
</tr>
<tr>
<td>Device support</td>
<td>Your system and LBM manage bandwidth for all types of devices, including IP phones, gateways, and H.323 and SIP trunk destinations. However, intercluster enhanced locations CAC requires SIP ICTs assigned to the system shadow location. All other types of devices are supported only when assigned to ordinary (fixed) locations.</td>
</tr>
</tbody>
</table>

Enhanced Locations Call Admission Control Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bandwidth Reservation Path</td>
<td>During network failure conditions, the bandwidth reservation path calculated by Unified Communications Manager might not accurately reflect network conditions. There is no satisfactory way to allow for this scenario in the model.</td>
</tr>
<tr>
<td>Bandwidth and Video Capabilities</td>
<td>If video capabilities are enabled, then bandwidth for audio will be allocated from video.</td>
</tr>
<tr>
<td>Synchronization</td>
<td>The model created by the system is not perfectly synchronized at all times. Use conservative bandwidth allocations to accommodate this restriction.</td>
</tr>
</tbody>
</table>
CHAPTER 30

Configure Resource Reservation Protocol

- RSVP Call Admission Control Overview, on page 213
- RSVP Call Admission Control Prerequisites, on page 213
- RSVP Configuration Task Flow, on page 213

RSVP Call Admission Control Overview

Resource Reservation Protocol (RSVP) is a resource-reservation, transport-level protocol for reserving resources in IP networks. You can use RSVP as an alternative to enhanced-locations call admission control (CAC). RSVP reserves resources for specific sessions. A session is a flow that has a particular destination address, destination port, and a protocol identifier (TCP or UDP).

RSVP Call Admission Control Prerequisites

You must use IPv4 addressing. RSVP does not support IPv6 addressing.

RSVP Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Clusterwide Default RSVP Policy, on page 214</td>
<td>Configure the RSVP policy for all nodes in the cluster.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Location-pair RSVP Policy, on page 215</td>
<td>Optional. You can configure the RSVP policy for a specific location pair if you want the location pair to use a different policy than the rest of the cluster.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure RSVP Retry, on page 216</td>
<td>Configure the frequency and number of RSVP retries.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Midcall RSVP Error Handling, on page 216</td>
<td>Configure how the system responds when RSVP fails during a call.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>------------------------------------------------------------</td>
<td>-------------------------------------------------------------------------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure MLPP-to-RSVP Priority Mapping, on page 217</td>
<td>Optional. If you use multilevel precedence and preemption (MLPP), map the caller MLPP precedence level to an RSVP priority.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> Configure RSVP agents.</td>
<td>Perform this IOS procedure on your gateway device. See the documentation for device information about how to configure an RSVP agent.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> Configure the Application ID, on page 218</td>
<td>When you configure the RSVP application ID, the system adds an identifier to both the voice and video traffic so that the Cisco RSVP Agent can set a separate bandwidth limit on either type of traffic, based on the identifier it receives.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> Configure DSCP Marking, on page 218</td>
<td>Configure DSCP marking so that if the RSVP reservation fails, the system can instruct the RSVP agent or endpoint devices to change media Differentiated Services Control Point (DSCP) marking to best effort. Otherwise, an excess of EF-marked media packets can degrade quality of service (QoS) even for flows that have a reservation.</td>
<td></td>
</tr>
</tbody>
</table>

## Configure Clusterwide Default RSVP Policy

Configure the RSVP policy for all nodes in the cluster.

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>In Cisco Unified Communications Manager Administration, choose <strong>System &gt; Service Parameters</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>In the <strong>Service Parameter Configuration</strong> window, choose a server and choose the Cisco CallManager service.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>In the <strong>Clusterwide Parameters (System - RSVP)</strong> section, configure the Default Interlocation RSVP Policy service parameter.</td>
</tr>
</tbody>
</table>

You can set this service parameter to the following values:

- No Reservation-No RSVP reservations get made between any two locations.
- Optional (Video Desired)-A call can proceed as a best-effort, audio-only call if failure to obtain reservations for both audio and video streams occurs. RSVP agent continues to attempt RSVP reservation for audio and informs Cisco Unified Communications Manager if reservation succeeds.
- Mandatory-Cisco Unified Communications Manager does not ring the terminating device until RSVP reservation succeeds for the audio stream and, if the call is a video call, for the video stream as well.
• Mandatory (Video Desired)-A video call can proceed as an audio-only call if a reservation for the audio stream succeeds but a reservation for the video stream does not succeed.

What to do next
Choose one of the following options:

• If you want a location pair to use a different policy than the rest of the cluster, Configure Location-pair RSVP Policy, on page 215.

• If you are using the same RSVP policy for all nodes in the cluster, Configure RSVP Retry, on page 216.

Configure Location-pair RSVP Policy

You can configure the RSVP policy for a specific location pair if you want the location pair to use a different policy than the rest of the cluster. When you use this procedure, the RSVP policy that you configure for the location pair overrides the policy that you configured for the cluster.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified Communications Manager Administration, choose the System &gt; Location.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Find one location of the location pair and select this location.</td>
</tr>
<tr>
<td>Step 3</td>
<td>To modify the RSVP policy between the selected location and another location, select the other location in the location pair.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the RSVP Setting drop-down list, choose an RSVP policy for this location pair.</td>
</tr>
</tbody>
</table>

You can set this field to the following values:

• Use System Default—The RSVP policy for the location pair matches the cluster-wide RSVP policy.

• No Reservation—No RSVP reservations get made between any two locations.

• Video Desired (Optional) –A call can proceed as a best-effort, audio-only call if failure to obtain reservations for both audio and video streams occurs. The RSVP agent continues to attempt RSVP reservation for audio and informs Cisco Unified Communications Manager if reservation succeeds. The system does not ring the terminating device until RSVP reservation succeeds for the audio stream and, if the call is a video call, for the video stream as well.

• Video Desired–A video call can proceed as an audio-only call if a reservation for the audio stream succeeds but the reservation for the video stream does not succeed.

What to do next
Configure RSVP Retry, on page 216
Configure RSVP Retry

Use this procedure to configure the frequency and number of RSVP retries.

**Before you begin**

- Configure Clusterwide Default RSVP Policy, on page 214
- Optional. Configure Location-pair RSVP Policy, on page 215

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified Communications Manager Administration, choose System &gt; Service Parameters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the Service Parameter Configuration window, choose a server and choose the Cisco CallManager service.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Clusterwide Parameters (System - RSVP) section, configure the specified service parameters.</td>
</tr>
</tbody>
</table>

You can set these service parameters to the following values:

- RSVP Retry Timer-Specify the RSVP retry timer value in seconds. If you set this parameter to 0, you disable RSVP retry on the system.

- Mandatory RSVP Midcall Retry Counter-Specify the midcall RSVP retry counter when the RSVP policy specifies Mandatory and midcall error handling option is set to “call fails following retry counter exceeds.” The default value specifies 1 time. If you set the service parameter to -1, retry continues indefinitely until either the reservation succeeds or the call gets torn down.

**What to do next**

Configure Midcall RSVP Error Handling, on page 216

Configure Midcall RSVP Error Handling

Use this procedure to configure midcall RSVP error handling.

**Before you begin**

Configure RSVP Retry, on page 216

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified Communications Manager Administration, choose System &gt; Service Parameters.</th>
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</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the Service Parameter Configuration window, choose a server and choose the Cisco CallManager service.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Clusterwide Parameters (System - RSVP) section, configure the specified service parameter.</td>
</tr>
</tbody>
</table>

You can set the Mandatory RSVP mid call error handle option service parameter to the following values:
• Call becomes best effort-If RSVP fails during a call, the call becomes a best-effort call. If retry is enabled, RSVP retry attempts begin simultaneously.

• Call fails following retry counter exceeded-If RSVP fails during a call, the call fails after N retries of RSVP, where the Mandatory RSVP Mid-call Retry Counter service parameter specifies N.

What to do next

Configure RSVP agents on your gateway device. See the documentation for device for information about how to configure an RSVP agent. After you have configure RSVP agents on your gateway, return to Cisco Unified Communications Manager Administration and choose one of the following options:

• Optional. Configure MLPP-to-RSVP Priority Mapping, on page 217 if you are using multilevel precedence and preemption in your network.

• Configure the Application ID, on page 218

Configure MLPP-to-RSVP Priority Mapping

Optional. Use the following clusterwide (System - RSVP) service parameters to configure the mapping from a caller MLPP precedence level to RSVP priority:

• MLPP EXECUTIVE OVERRIDE To RSVP Priority Mapping
• MLPP FLASH OVERRIDE To RSVP Priority Mapping
• MLPP FLASH To RSVP Priority Mapping
• MLPP IMMEDIATE To RSVP Priority Mapping
• MLPP PL PRIORITY To RSVP Priority Mapping
• MLPP PL ROUTINE To RSVP Priority Mapping

To locate and configure these service parameters, follow these steps:

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose System > Service Parameters.

Step 2 In the Service Parameter Configuration window, choose a server and choose the Cisco CallManager service.

Step 3 In the Clusterwide Parameters (System - RSVP) section, configure the specified service parameters.

These service parameters function as follows:

• Cisco Unified Communications Manager maps the caller precedence level to RSVP priority when initiating an RSVP reservation based on the following configuration: the higher the service parameter value, the higher the priority.

• The IOS router preempts the call based on RSVP priority.

• The RSVP agent must notify Cisco Unified Communications Manager about the reason for an RSVP reservation failure, including the cause for preemption.
Cisco Unified Communications Manager uses the existing MLPP mechanism to notify the preempted calling and called parties about the preemption.

What to do next
Configure RSVP agents on your gateway device. See the documentation for device for information about how to configure an RSVP agent. After you have configure RSVP agents on your gateway, return to Cisco Unified Communications Manager Administration and Configure the Application ID, on page 218.

Configure the Application ID

When you configure the RSVP application ID, the system adds an identifier to both the voice and video traffic so that the Cisco RSVP Agent can set a separate bandwidth limit on either type of traffic, based on the identifier it receives.

Before you begin this procedure, configure RSVP agents on your gateway device. See the documentation for device for information about how to configure an RSVP agent.

Before you begin
To deploy the RSVP application ID in the network, you must use a minimum version of Cisco IOS Release 12.4(6)T or higher on the Cisco RSVP Agent router.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
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<tbody>
<tr>
<td>1</td>
<td>In Cisco Unified Communications Manager Administration, choose System &gt; Service Parameters.</td>
</tr>
<tr>
<td>2</td>
<td>In the Service Parameter Configuration window, choose a server and choose the Cisco CallManager service.</td>
</tr>
<tr>
<td>3</td>
<td>In the Clusterwide Parameters (System - RSVP) section, configure the RSVP Audio Application ID service parameter. (Default = AudioStream)</td>
</tr>
<tr>
<td>4</td>
<td>In the Clusterwide Parameters (System - RSVP) section, configure the RSVP Video Application ID (Default = VideoStream)</td>
</tr>
</tbody>
</table>

What to do next
Configure DSCP Marking, on page 218

Configure DSCP Marking

If the RSVP reservation fails, the system instructs the RSVP agent or endpoint devices (in case a failure to allocate an RSVP agent occurs) to change media Differentiated Services Control Point (DSCP) marking to best effort. Otherwise, an excess of EF-marked media packets can degrade quality of service (QoS) even for flows that have a reservation.
Before you begin
Configure the Application ID, on page 218

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Procedure</th>
</tr>
</thead>
<tbody>
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<td><strong>Step 1</strong></td>
<td>In Cisco Unified Communications Manager Administration, choose <strong>System &gt; Service Parameters</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>In the <strong>Service Parameter Configuration</strong> window, choose a server and choose the Cisco CallManager service.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>In the <strong>Clusterwide Parameters (System - QoS)</strong> section, configure the <strong>DSCP for Audio Calls When RSVP Fails</strong> service parameter.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>In the <strong>Clusterwide Parameters (System - QoS)</strong> section, configure the <strong>DSCP for Video Calls When RSVP Fails</strong> service parameter.</td>
</tr>
</tbody>
</table>
Configure DSCP Marking
Configure End Users

• End User Configuration Overview, on page 223
• Configure User Access, on page 227
• Configure Credential Policy, on page 245
• Configure User Profiles, on page 249
• Configure Service Profile, on page 253
• Configure Feature Group Template, on page 261
• Import Users From LDAP Directory, on page 263
• Provision End Users Manually, on page 275
End User Configuration Overview

• About End User Configuration, on page 223
• End User Configuration, on page 223

About End User Configuration

The chapters in this part describe how to provision and configure end users for your system. End users are the main consumers of Cisco Unified Communications Manager features. End users can be assigned with phones and directory numbers thereby allowing your end users to make calls and communicate with other users in the system as well as placing calls to external networks such as the PSTN.

For provisioning large numbers of end users at once, Cisco Unified Communications Manager provides the following features:

• LDAP Directory Integration—You can synchronize Cisco Unified Communications Manager with an external LDAP directory thereby allowing you to import end user data from the LDAP directory.

• Bulk Administration Tool—You can use the Bulk Administration Tool to import and configure a large number of end users, and associated user data, from a CSV file in a single operation.

After your end users are provisioned, you can configure user settings such as user profiles that allow your users to provision their own phones, in addition to phone services and credential policies.

End User Configuration

Complete the following task flows to configure end users for your system.

Procedure

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<th>Purpose</th>
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<tbody>
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<td>Step 1</td>
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<tr>
<td>Command or Action</td>
<td>Purpose</td>
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<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure credential policies for your end users.</td>
</tr>
<tr>
<td><strong>Credential Policy Configuration Task Flow, on page 246</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure user profiles for groups of users who meet the same access and functionality requirements. The user profile consists of common phone and phone line settings that can allow you to quickly configure new phones and phone lines for users who use this user profile. You can also enable self-provisioning for users who use this profile.</td>
</tr>
<tr>
<td><strong>UserProfile Configuration Task Flow, on page 250</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure a service profile with settings for Unified Communications (UC) services. You can apply the service profile to groups of users who have the same services requirements. The service profile allows you to configure UC services for any new phones provisioned for users who use this service profile.</td>
</tr>
<tr>
<td><strong>Service Profile Configuration Task Flow, on page 254</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Optional. Configure a feature group template to your end users. A feature group template contains a set of common feature configurations as well as an assigned user profile and service profile. For LDAP synchronized users you can assign the feature group template during the LDAP sync thereby assigning a user profile, service profile, line and device templates, and self-provisioning capability to the user.</td>
</tr>
<tr>
<td><strong>Configure a Feature Group Template, on page 262</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>If your deployment includes a company LDAP directory, you can import your end users directly from the LDAP directory into the Cisco Unified Communications Manager database.</td>
</tr>
<tr>
<td><strong>LDAP Synchronization Configuration Task Flow, on page 265</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>If you are not importing end users from an LDAP directory, you can use the Bulk Administration Tool to import the end user list, and end user configurations, from a CSV file into the Cisco Unified Communications Manager database. For details on how to use the Bulk Administration Guide to perform bulk transactions to the database, see the Bulk Administration Guide for Cisco Unified Communications Manager at <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html">http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html</a>.</td>
</tr>
<tr>
<td><strong>LDAP Synchronization Configuration Task Flow, on page 265</strong></td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
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<tr>
<td>--------------------------------------------------------</td>
<td>-------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Step 8</strong> Manual End User Configuration Task Flow, on page 276</td>
<td>Optional. Add new users to the database manually.</td>
</tr>
</tbody>
</table>
Configure User Access

- User Access Overview, on page 227
- User Access Prerequisites, on page 229
- User Access Configuration Task Flow, on page 230
- Standard Roles and Access Control Groups, on page 235

User Access Overview

Manage user access to Cisco Unified Communications Manager by configuring the following items:

- Access Control Groups
- Roles
- User Rank

Roles Overview

Users obtain system access privileges via the roles that are associated to the access control group of which the user is a member. Each role contains a set of permissions that is attached to a specific resource or application, such as Cisco Unified CM Administration or CDR Analysis and Reporting. For an application such as Cisco Unified CM Administration, the role may contain permissions that let you view or edit specific GUI pages in the application. There are three levels of permissions that you can assign to a resource or application:

- Read—Allows a user to view settings for a resource.
- Update—Allows a user to edit settings for a resource.
- No Access—If a user has neither Read or Update access, the user has no access to view or edit settings for a given resource.

Role Types

When provisioning users, you must decide what roles you want to apply and then assign users to an access control group that contains the role. There are two main types of roles in Cisco Unified Communications Manager:
• Standard roles—These are preinstalled default roles that are designed to meet the needs of common deployments. You cannot edit permissions for standard roles.

• Custom roles—Create custom roles when no standard roles have the privileges you need.

**Access Control Group Overview**

An access control group is a list of users and the roles that are assigned to those users. When you assign an end user, application user, or administrator user to an access control group, the user gains the access permissions of the roles that are associated to the group. You can manage system access by assigning users with similar access needs to an access control group with only the roles and permissions that they need.

There are two types of access control groups:

• **Standard Access Control Groups**—These are predefined default groups with role assignments that meet common deployment needs. You cannot edit the role assignments in a standard group. However, you can add and delete users, in addition to editing the User Rank requirement. For a list of standard access control groups, and their associated roles, see Standard Roles and Access Control Groups, on page 235.

• **Custom Access Control Groups**—Create your own access control groups when none of the standard groups contain the role permissions that meet your needs.

The User Rank framework provides a set of controls over the access control groups to which a user can be assigned. To be assigned to an access control group, a user must meet the minimum rank requirement for that group. For example, end users whom have a User Rank of 4 can be assigned only to access control groups with minimum rank requirements between 4 and 10. They cannot be assigned to groups with a minimum rank of 1.

**Example - Role Permissions with Access Control Groups**

The following example illustrates a cluster where the members of a testing team are assigned to access control group **test_ACG**. The screen capture on the right displays the access settings of test_Role, which is the role that is associated to the access control group. Also note that the access control group has a minimum rank requirement of 3. All of the group members must have a rank between 1-3 to be able to join the group.
User Rank Overview

The User Rank hierarchy provides a set of controls over which access control groups an administrator can assign to an end user or application user.

When provisioning end users or application users, administrators can assign a user rank for the user. Administrators can also assign a user rank requirement for each access control group. When adding users to access control groups, administrators can assign users only to the groups where the user's User Rank meets the group's rank requirement. For example, an administrator can assign a user whom has a User Rank of 3 to access control groups that have a User Rank requirement between 3 and 10. However, an administrator cannot assign that user to an access control group that has a User Rank requirement of 1 or 2.

Administrators can create their own user rank hierarchy within the User Rank Configuration window and can use that hierarchy when provisioning users and access control groups. Note that if you don't configure a user rank hierarchy, or if you simply don't specify the User Rank setting when provisioning users or access control groups, all users and access control groups are assigned the default User Rank of 1 (the highest rank possible).

User Access Prerequisites

Before you provision end users:

- Standard Roles and Access Control Groups, on page 235—Review the list of predefined roles and access control groups. Determine if you will need to configure customized roles and groups.
• Plan which user ranks you will assign to your users and groups.

## User Access Configuration Task Flow

### Procedure

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<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
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<td>Configure User Rank Hierarchy, on page 230</td>
<td>Set up the user rank hierarchy for your system.</td>
</tr>
<tr>
<td>Step 2</td>
<td>If you need to create a new role, use one of the following methods:</td>
<td>Use the 'Create' procedure if you want to create and configure a new role from scratch. Use the 'Copy' procedure if the new role has similar privileges to an existing role. You can copy the privileges from the existing role into a new role, and then make edits to the privileges in the new role.</td>
</tr>
<tr>
<td></td>
<td>• Create a Custom Role, on page 231</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Copy an Existing Role, on page 232</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>If you need to create new access control groups, use one of the following methods:</td>
<td>Use the 'Create' procedure to create a new access control group from scratch. Use the 'Copy' procedure if the new access control group has similar settings to an existing access control group. You can copy the settings from the existing access control group into a new group and then edit the settings.</td>
</tr>
<tr>
<td></td>
<td>• Create Access Control Groups, on page 232</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Copy Access Control Group, on page 233</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Assign Roles to Access Control Group, on page 234</td>
<td>If you created a new access control group, assign roles to your access control group.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure Overlapping Privilege Policy, on page 235</td>
<td>Configure an enterprise policy to cover overlapping access privileges. This covers the situation where end users or application users are assigned to multiple access control groups or roles, each with conflicting privilege settings.</td>
</tr>
</tbody>
</table>

### Related Topics
- Standard Roles and Access Control Groups, on page 235

## Configure User Rank Hierarchy

Use this procedure to create a custom user rank hierarchy.

### Note

If you don't configure a user rank hierarchy, all users and access control groups get assigned a user rank of 1 (the highest possible rank) by default.
Create a Custom Role

Create a custom role when there is no system-defined role with the privilege settings that you require.

Tip
If the privileges in the new role that you want to create are similar to that of an existing role, follow the procedure Copy an Existing Role, on page 232 to copy the existing privileges into a new role that you can edit.

Procedure

Step 1 In Cisco Unified CM Administration, click User Management > User Settings > Role.
Step 2 Do either of the following:
   • To create a new role, click Add New. Choose the Application with which this role associates, and click Next.
   • To copy settings from an existing role, click Find and open the existing role. Click Copy and enter a name for the new role. Click OK.
Step 3 Enter a Name and Description for the role.
Step 4 For each resource, check the boxes that apply:
   • Check the Read check box if you want users to be able to view settings for the resource.
   • Check the Update check box if you want users to be able to edit settings for the resource.
   • Leave both check boxes unchecked to provide no access to the resource.
Step 5 Click Grant access to all or Deny access to all button to grant or remove privileges to all resources that display on a page for this role.
   Note If the list of resources displays on more than one page, this button applies only to the resources that display on the current page. You must display other pages and use the button on those pages to change the access to the resources that are listed on those pages.
Step 6 Click Save.
Copy an Existing Role

The Copy command allows you to create new roles that are based on the settings of existing roles. Cisco Unified Communications Manager does not allow you to edit standard roles, but you can use the Copy command to create a new role with the identical resources and privileges as the standard role. You can then edit the privileges in the new role that you created.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, click User Management > User Settings > Role.
Step 2 Click Find and select the role whose resources and privileges you want to copy.
Step 3 Click Copy.
Step 4 Enter the name of the new role and click OK. The Role Configuration window displays the settings of the new role. The privileges for the new role are the same as the privileges for the role you copied.
Step 5 For any of the resources in the new role, edit the privileges as follows:

• Check the Read check box to allow users to view the resource.
• Check the Update check box to allow users to edit the resource.
• To restrict access to the resource, leave both check boxes unchecked.

Step 6 Click Save.

What to do next

Create a new access control group using one of the following methods:

• Create Access Control Groups, on page 232
• Copy Access Control Group, on page 233

Related Topics

Standard Roles and Access Control Groups, on page 235

Create Access Control Groups

Use this procedure if you need to create a new access control group. You may need to create a new access control group if the system-defined access control groups do not meet your deployment needs.

Before you begin

If you need to create new roles, perform one of the following procedures:

• Create a Custom Role, on page 231
Copy an Existing Role, on page 232

Procedure

Step 1 In Cisco Unified CM Administration, choose User Management > User Settings > Access Control Groups.
Step 2 Click Add New.
Step 3 Enter a Name for the access control group.
Step 4 From the Available for Users with User Rank as drop-down, select the minimum User Rank for a user to be assigned to this group. The default user rank is 1.
Step 5 Click Save.

What to do next
Assign Roles to Access Control Group, on page 234

Copy Access Control Group

Create a custom access control group by copying the settings from an existing access control group. When you copy an existing access control group, the system copies all the settings, including any assigned roles and users, to the new access control group. However, unlike default access control groups, you can make edits to the roles assigned to a custom access control group.

Before you begin
If you need to create a new role, perform either of the following steps:

• Create a Custom Role, on page 231
• Copy an Existing Role, on page 232

Procedure

Step 1 In Cisco Unified CM Administration, choose User Management > User Settings > Access Control Groups.
Step 2 Click Find and select the access control group whose settings you want to copy.
Step 3 Click Copy.
Step 4 Enter a name for the new access control group and click OK.
Step 5 From the Available for Users with User Rank as drop-down, select the minimum User Rank for a user to be assigned to this group.
Step 6 Click Save.

What to do next
Assign Roles to Access Control Group, on page 234
Assign Roles to Access Control Group

For any new access control groups that you create, assign roles to the access control group. If you copied the access control group from an existing group, you may also need to delete a role.

**Note**

You cannot edit the role assignments for any of the standard access control groups that are configured by default.

**Before you begin**

Perform either of the following tasks to create a new access control group:

- Create Access Control Groups, on page 232
- Copy Access Control Group, on page 233

**Procedure**

**Step 1**
In Cisco Unified CM Administration, choose **User Management > User Settings > Access Control Group**.

**Step 2**
Click **Find** and select an access control group.

**Step 3**
From the **Related Links** drop-down list box, select **Assign Role to Access Control Group** and click **Go**.

**Step 4**
If you need to assign a role:
   a) Click **Assign Role to Group**.
   b) In the **Find and List Roles** window, check the roles that you want to assign to the group.
   c) Click **Add Selected**.

**Step 5**
If you need to delete a role:
   a) In the **Role** list box, highlight the role that you want to delete.
   b) Click **Delete Role Assignment**.

**Step 6**
Click **Save**.

**What to do next**

Configure Overlapping Privilege Policy, on page 235
Configure Overlapping Privilege Policy

Configure how Cisco Unified Communications Manager handles overlapping user privileges in access control group assignments. This is to cover situations where an end user is assigned to multiple access control groups, each with conflicting roles and access privileges.

**Before you begin**

Assign Roles to Access Control Group, on page 234

**Procedure**

**Step 1**

In Cisco Unified CM Administration, choose **System > Enterprise Parameters**.

**Step 2**

Under **User Management Parameters**, configure one of the following values for the **Effective Access Privileges For Overlapping User Groups and Roles** as follows:

- **Maximum**—The effective privilege represents the maximum of the privileges of all the overlapping access control groups. This is the default option.
- **Minimum**—The effective privilege represents the minimum of the privileges of all the overlapping access control groups.

**Step 3**

Click **Save**.

---

**Standard Roles and Access Control Groups**

The following table summarizes the standard roles and access control groups that come preconfigured on Cisco Unified Communications Manager. The privileges for a standard role are configured by default. In addition, the access control groups that are associated with a standard role are also configured by default.

For both standard roles and the associated access control group, you cannot edit any of the privileges, or the role assignments.

<table>
<thead>
<tr>
<th>Standard Role</th>
<th>Privileges/Resources for the Role</th>
<th>Associated Standard Access Control Group(s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standard AXL API Access</td>
<td>Allows access to the AXL database API</td>
<td>Standard CCM Super Users</td>
</tr>
<tr>
<td>Standard AXL API Users</td>
<td>Grants login rights to execute AXL APIs.</td>
<td></td>
</tr>
<tr>
<td>Standard AXL Read Only API Access</td>
<td>Allows you to execute AXL read only APIs (list APIs, get APIs, executeSQLQuery API) by default.</td>
<td></td>
</tr>
<tr>
<td>Standard Admin Rep Tool Admin</td>
<td>Allows you to view and configure Cisco Unified Communications Manager CDR Analysis and Reporting (CAR).</td>
<td>Standard CAR Admin Users, Standard CCM Super Users</td>
</tr>
<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
</tr>
<tr>
<td>----------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------</td>
<td>---------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Standard Audit Log Administration</td>
<td>Allows you to perform the following tasks for the audit logging feature:</td>
<td>Standard Audit Users</td>
</tr>
<tr>
<td></td>
<td>• View and configure audit logging in the Audit Log Configuration window in Cisco Unified Serviceability</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• View and configure trace in Cisco Unified Serviceability and collect traces for the audit log feature in the Real-Time Monitoring Tool</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• View and start/stop the Cisco Audit Event service in Cisco Unified Serviceability</td>
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<tr>
<td></td>
<td>• View and update the associated alert in the RTMT</td>
<td></td>
</tr>
<tr>
<td>Standard CCM End Users</td>
<td>Grant an end user log-in rights to the Cisco Unified Communications Self Care Portal</td>
<td>Standard CCM End Users</td>
</tr>
<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
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<tr>
<td>---------------------------------------------</td>
<td>--------------------------------------------------------------------------------------------------</td>
<td>-------------------------------------------------------------------</td>
</tr>
<tr>
<td>Standard CCM Feature Management</td>
<td>Allows you to perform the following tasks in Cisco Unified Communications Manager Administration:</td>
<td>Standard CCM Server Maintenance</td>
</tr>
<tr>
<td></td>
<td>• View, delete, and insert the following items by using the Bulk Administration Tool:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Client matter codes and forced authorization codes</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Call pickup groups</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• View and configure the following items in Cisco Unified Communications Manager Administration:</td>
<td></td>
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<tr>
<td></td>
<td>• Client matter codes and forced authorization codes</td>
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<td></td>
<td>• Call park</td>
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<tr>
<td></td>
<td>• Call pickup</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Meet-Me numbers/patterns</td>
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<tr>
<td></td>
<td>• Message Waiting</td>
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<td></td>
<td>• Cisco Unified IP Phone Services</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Voice mail pilots, voice mail port wizard, voice mail ports, and voice mail profiles</td>
<td></td>
</tr>
<tr>
<td>Standard CCM Gateway Management</td>
<td>Allows you to perform the following tasks in Cisco Unified Communications Manager Administration:</td>
<td>Standard CCM Gateway Administration</td>
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<tr>
<td></td>
<td>• View and configure gateway templates in the Bulk Administration Tool</td>
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</tr>
<tr>
<td></td>
<td>• View and configure gatekeepers, gateways, and trunks</td>
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<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
</tr>
<tr>
<td>--------------------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
<td>------------------------------------------------------------------</td>
</tr>
<tr>
<td>Standard CCM Phone Management</td>
<td>Allows you to perform the following tasks in Cisco Unified Communications Manager Administration:</td>
<td>Standard CCM Phone Administration</td>
</tr>
<tr>
<td></td>
<td>• View and export phones in the Bulk Administration Tool</td>
<td></td>
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<tr>
<td></td>
<td>• View and insert user device profiles in the Bulk Administration Tool</td>
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<tr>
<td></td>
<td>• View and configure the following items in Cisco Unified Communications Manager Administration:</td>
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</tr>
<tr>
<td></td>
<td>• BLF speed dials</td>
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</tr>
<tr>
<td></td>
<td>• CTI route points</td>
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</tr>
<tr>
<td></td>
<td>• Default device profiles or default profiles</td>
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</tr>
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<td></td>
<td>• Directory numbers and line appearances</td>
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<td></td>
<td>• Firmware load information</td>
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<tr>
<td></td>
<td>• Phone button templates or softkey templates</td>
<td></td>
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<tr>
<td></td>
<td>• Phones</td>
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</tr>
<tr>
<td></td>
<td>• Reorder phone button information for a particular phone by clicking the Modify Button Items button in the Phone Configuration window</td>
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</tr>
<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
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<tr>
<td>Standard CCM Route Plan Management</td>
<td>Allows you to perform the following tasks in Cisco Unified Communications Manager Administration:</td>
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<td>• View and configure application dial rules</td>
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<tr>
<td></td>
<td>• View and configure calling search spaces and partitions</td>
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</tr>
<tr>
<td></td>
<td>• View and configure dial rules, including dial rule patterns</td>
<td></td>
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<tr>
<td></td>
<td>• View and configure hunt lists, hunt pilots, and line groups</td>
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<td></td>
<td>• View and configure route filters, route groups, route hunt list, route lists, route patterns, and route plan report</td>
<td></td>
</tr>
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<td></td>
<td>• View and configure time period and time schedule</td>
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<td></td>
<td>• View and configure translation patterns</td>
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<tr>
<td>Standard CCM Service Management</td>
<td>Allows you to perform the following tasks in Cisco Unified Communications Manager Administration:</td>
<td>Standard CCM Server Maintenance</td>
</tr>
<tr>
<td></td>
<td>• View and configure the following items:</td>
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<td></td>
<td>• Annunciators, conference bridges, and transcoders</td>
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<td>• audio sources and MOH servers</td>
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<td>• Media resource groups and media resource group lists</td>
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<td>• Media termination point</td>
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<td></td>
<td>• Cisco Unified Communications Manager Assistant wizard</td>
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<td></td>
<td>• View and configure the Delete Managers, Delete Managers/Assistants, and Insert Managers/Assistants windows in the Bulk Administration Tool</td>
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</table>
## Standard Roles and Access Control Groups

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<th>Privileges/Resources for the Role</th>
<th>Associated Standard Access Control Group(s)</th>
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<td>Standard CCM System Management</td>
<td>Allows you to perform the following tasks in Cisco Unified Communications Manager Administration:</td>
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<td>• View and configure the following items:</td>
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<td>• Automate Alternate Routing (AAR) groups</td>
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<tr>
<td></td>
<td>• Cisco Unified Communications Managers (Cisco Unified CMs) and Cisco Unified Communications Manager groups</td>
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<tr>
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<td>• Date and time groups</td>
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<tr>
<td></td>
<td>• Device defaults</td>
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<td>• Device pools</td>
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<td>• Enterprise parameters</td>
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<td>• Enterprise phone configuration</td>
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<td>• Locations</td>
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<td></td>
<td>• Network Time Protocol (NTP) servers</td>
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<tr>
<td></td>
<td>• Plug-ins</td>
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<tr>
<td></td>
<td>• Security profiles for phones that run Skinny Call Control Protocol (SCCP) or Session Initiation Protocol (SIP); security profiles for SIP trunks</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Survivable Remote Site Telephony (SRST) references</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Servers</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• View and configure the Job Scheduler windows in the Bulk Administration Tool</td>
<td></td>
</tr>
<tr>
<td>Standard CCM User Privilege Management</td>
<td>Allows you to view and configure application users in Cisco Unified Communications Manager Administration.</td>
<td></td>
</tr>
<tr>
<td>Standard CCMADMIN Administration</td>
<td>Allows you access to all aspects of the CCMAdmin system</td>
<td></td>
</tr>
<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
</tr>
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<td>---------------------------------------------</td>
</tr>
<tr>
<td>Standard CCMADMIN Administration</td>
<td>Allows you to view and configure all items in Cisco Unified Communications Manager Administration and the Bulk Administration Tool.</td>
<td>Standard CCM Super Users</td>
</tr>
<tr>
<td>Standard CCMADMIN Administration</td>
<td>Allows you to view and configure information in the Dialed Number Analyzer.</td>
<td></td>
</tr>
<tr>
<td>Standard CCMADMIN Read Only</td>
<td>Allows read access to all CCMAdmin resources</td>
<td>Standard CCM Gateway Administration, Standard CCM Phone Administration, Standard CCM Read Only, Standard CCM Server Maintenance, Standard CCM Server Monitoring</td>
</tr>
<tr>
<td>Standard CCMADMIN Read Only</td>
<td>Allows you to view configurations in Cisco Unified Communications Manager Administration and the Bulk Administration Tool.</td>
<td></td>
</tr>
<tr>
<td>Standard CCMADMIN Read Only</td>
<td>Allows you to analyze routing configurations in the Dialed Number Analyzer.</td>
<td></td>
</tr>
<tr>
<td>Standard CCMUSER Administration</td>
<td>Allows access to the Cisco Unified Communications Self Care Portal.</td>
<td>Standard CCM End Users</td>
</tr>
<tr>
<td>Standard CTI Allow Call Monitoring</td>
<td>Allows CTI applications/devices to monitor calls</td>
<td>Standard CTI Allow Call Monitoring</td>
</tr>
<tr>
<td>Standard CTI Allow Call Park Monitoring</td>
<td>Allows CTI applications/devices to use call park</td>
<td>Standard CTI Allow Call Park Monitoring</td>
</tr>
<tr>
<td>Standard CTI Allow Call Recording</td>
<td>Allows CTI applications/devices to record calls</td>
<td>Standard CTI Allow Call Recording</td>
</tr>
<tr>
<td>Standard CTI Allow Calling Number Modification</td>
<td>Allows CTI applications to transform calling party numbers during a call</td>
<td>Standard CTI Allow Calling Number Modification</td>
</tr>
<tr>
<td>Standard CTI Allow Control of All Devices</td>
<td>Allows control of all CTI-controllable devices</td>
<td>Standard CTI Allow Control of All Devices</td>
</tr>
<tr>
<td>Standard CTI Allow Control of Phones Supporting Connected Xfer and conf</td>
<td>Allows control of all CTI devices that supported connected transfer and conferencing</td>
<td>Standard CTI Allow Control of Phones supporting Connected Xfer and conf</td>
</tr>
<tr>
<td>Standard CTI Allow Control of Phones Supporting Rollover Mode</td>
<td>Allows control of all CTI devices that supported Rollover mode</td>
<td>Standard CTI Allow Control of Phones supporting Rollover Mode</td>
</tr>
<tr>
<td>Standard CTI Allow Reception of SRTP Key Material</td>
<td>Allows CTI applications to access and distribute SRTP key material</td>
<td>Standard CTI Allow Reception of SRTP Key Material</td>
</tr>
<tr>
<td>Standard CTI Enabled</td>
<td>Enables CTI application control</td>
<td>Standard CTI Enabled</td>
</tr>
<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
</tr>
<tr>
<td>---------------------------------------------------</td>
<td>--------------------------------------------------------------------------------------------------</td>
<td>----------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Standard CTI Secure Connection</td>
<td>Enables a secure CTI connection to Cisco Unified Communications Manager</td>
<td>Standard CTI Secure Connection</td>
</tr>
<tr>
<td>Standard CUReporting</td>
<td>Allows application users to generate reports from various sources</td>
<td></td>
</tr>
<tr>
<td>Standard CUReporting</td>
<td>Allows you to view, download, generate, and upload reports in Cisco Unified Reporting</td>
<td>Standard CCM Administration Users, Standard CCM Super Users</td>
</tr>
<tr>
<td>Standard EM Authentication Proxy Rights</td>
<td>Manages Cisco Extension Mobility (EM) authentication rights for applications; required for all application users that interact with Cisco Extension Mobility (for example, Cisco Unified Communications Manager Assistant and Cisco Web Dialer)</td>
<td>Standard CCM Super Users, Standard EM Authentication Proxy Rights</td>
</tr>
<tr>
<td>Standard Packet Sniffing</td>
<td>Allows you to access Cisco Unified Communications Manager Administration to enable packet sniffing (capturing).</td>
<td>Standard Packet Sniffer Users</td>
</tr>
<tr>
<td>Standard RealtimeAndTraceCollection</td>
<td>Allows you to access Cisco Unified Serviceability and the Real-Time Monitoring Tool view and use the following items:</td>
<td>Standard RealtimeAndTraceCollection</td>
</tr>
<tr>
<td></td>
<td>• Simple Object Access Protocol (SOAP) Serviceability AXL APIs</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• SOAP Call Record APIs</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• SOAP Diagnostic Portal (Analysis Manager) Database Service</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• configure trace for the audit log feature</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• configure Real-Time Monitoring Tool, including collecting traces</td>
<td></td>
</tr>
<tr>
<td>Standard Role</td>
<td>Privileges/Resources for the Role</td>
<td>Associated Standard Access Control Group(s)</td>
</tr>
<tr>
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<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
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</tr>
</tbody>
</table>
| Standard SERVICEABILITY             | Allows you to view and configure the following windows in Cisco Unified Serviceability or the Real-Time Monitoring Tool:  
  - Alarm Configuration and Alarm Definitions (Cisco Unified Serviceability)  
  - Audit Trace (marked as read/view only)  
  - SNMP-related windows (Cisco Unified Serviceability)  
  - Trace Configuration and Troubleshooting of Trace Configuration (Cisco Unified Serviceability)  
  - Log Partition Monitoring  
  - Alert Configuration (RTMT), Profile Configuration (RTMT), and Trace Collection (RTMT)  
  Allows you to view and use the SOAP Serviceability AXL APIs, the SOAP Call Record APIs, and the SOAP Diagnostic Portal (Analysis Manager) Database Service.  
  For the SOAP Call Record API, the RTMT Analysis Manager Call Record permission is controlled through this resource.  
  For the SOAP Diagnostic Portal Database Service, the RTMT Analysis Manager Hosting Database access controlled thorough this resource. | Standard CCM Server Monitoring, Standard CCM Super Users              |
<p>| Standard SERVICEABILITY Administration | A serviceability administrator can access the Plugin window in Cisco Unified Communications Manager Administration and download plugins from this window.                                                                                                                                                                       |                                                                     |
| Standard SERVICEABILITY Administration | Allows you to administer all aspects of serviceability for the Dialed Number Analyzer.                                                                                                                                                                                                                                           |                                                                     |</p>
<table>
<thead>
<tr>
<th>Standard Role</th>
<th>Privileges/Resources for the Role</th>
<th>Associated Standard Access Control Group(s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standard SERVICEABILITY Administration</td>
<td>Allows you to view and configure all windows in Cisco Unified Serviceability and Real-Time Monitoring Tool. (Audit Trace supports viewing only.) Allows you to view and use all SOAP Serviceability AXL APIs.</td>
<td></td>
</tr>
<tr>
<td>Standard SERVICEABILITY Read Only</td>
<td>Allows you to view all serviceability-related data for components in the Dialed Number Analyzer.</td>
<td>Standard CCM Read Only</td>
</tr>
<tr>
<td>Standard SERVICEABILITY Read Only</td>
<td>Allows you to view configuration in Cisco Unified Serviceability and Real-Time Monitoring Tool. (excluding audit configuration window, which is represented by the Standard Audit Log Administration role) Allows you to view all SOAP Serviceability AXL APIs, the SOAP Call Record APIs, and the SOAP Diagnostic Portal (Analysis Manager) Database Service.</td>
<td></td>
</tr>
<tr>
<td>Standard System Service Management</td>
<td>Allows you to view, activate, start, and stop services in Cisco Unified Serviceability.</td>
<td></td>
</tr>
<tr>
<td>Standard SSO Config Admin</td>
<td>Allows you to administer all aspects of SAML SSO configuration</td>
<td></td>
</tr>
<tr>
<td>Standard Confidential Access Level Users</td>
<td>Allows you to access all the Confidential Access Level Pages</td>
<td>Standard Cisco Call Manager Administration</td>
</tr>
<tr>
<td>Standard CCMADMIN Administration</td>
<td>Allows you to administer all aspects of CCMAdmin system</td>
<td>Standard Cisco Unified CM IM and Presence Administration</td>
</tr>
<tr>
<td>Standard CCMADMIN Read Only</td>
<td>Allows read access to all CCMAdmin resources</td>
<td>Standard Cisco Unified CM IM and Presence Administration</td>
</tr>
<tr>
<td>Standard CUReporting</td>
<td>Allows application users to generate reports from various sources</td>
<td>Standard Cisco Unified CM IM and Presence Reporting</td>
</tr>
</tbody>
</table>
CHAPTER 33

Configure Credential Policy

- Credential Policy Overview, on page 245
- Credential Policy Configuration Task Flow, on page 246

Credential Policy Overview

Credential policies control the authentication process for resources in Cisco Unified Communications Manager. A credential policy defines password requirements and account lockout details such as failed login attempts, expiration periods and lockout durations for end user passwords, end user PINs, and application user passwords. Credential policies can be assigned broadly to all accounts of a specific credential types, such as all end user PINs, or they can be customized for a specific application user, or end user.

Credential Types

In Credential Policy Configuration you can configure a new credential policy and then apply that new policy as the default credential policy for each of the following three credential types:

- End User PINs
- End User Passwords
- Application User Passwords

You can also apply the credential policy to a specific end user PIN, end user password, or application user password.

Credential Policies with LDAP Authentication Enabled

If your system is configured for LDAP Authentications with the corporate directory:

- With LDAP Authentication enabled, credential policies do not apply to end user passwords.
- Credential policies do apply to end user PINs and application user passwords, irrespective of whether LDAP Authentication is enabled. These password types use local authentication.

Note

Credential policies do not apply to operating system users or CLI users. These administrators use standard password verification procedures that the operating system supports.
Trivial Passwords

The system can be configured to check for trivial passwords and PINs. A trivial password is a credential that can be easily hacked, such as a password that be guessed easily such as using ABCD as your password or 123456 as your PIN.

Non-trivial passwords meet the following requirements:

- Must contain three of the following four characteristics: uppercase character, lowercase character, number, or symbol.
- Must not use a character or number more than three times consecutively.
- Must not repeat or include the alias, username, or extension.
- Cannot consist of consecutive characters or numbers. For example, passwords such as 654321 or ABCDEFG are not allowed.

PINs can contain digits (0-9) only. A non-trivial PIN meets the following criteria:

- Must not use the same number more than two times consecutively.
- Must not repeat or include the user extension, mailbox, or the reverse of the user extension or mailbox.
- Must contain three different numbers. For example, a PIN such as 121212 is trivial.
- Must not match the numeric representation (that is, dial by name) for the first or last name of the user.
- Must not contain groups of repeated digits, such as 408408, or patterns that are dialed in a straight line on a keypad, such as 2580, 159, or 753.

Credential Policy Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
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<tbody>
<tr>
<td><strong>Command or Action</strong></td>
</tr>
<tr>
<td><strong>Step 1</strong></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
</tr>
</tbody>
</table>

Related Topics

- Apply Credential Policy to End User, on page 277
Configure a Credential Policy

Configure a credential policy that you can apply as the default credential policy for all credentials that match a specific credential type such as end user PINs or end user passwords.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose User Management > Credential Policy.

**Step 2**
Perform one of the following steps:
- Click **Find** and select an existing credential policy.
- Click **Add New** to create a new credential policy.

**Step 3**
Complete the fields in the **Credential Policy Configuration** window. See the online help for more information about the fields and their configuration settings.

**Step 4**
Click **Save**.

**What to do next**
Configure Default Credentials for a Credential Policy, on page 247

Configure Default Credentials for a Credential Policy

Perform these steps to configure the default credentials for your credential policy. You can assign default credentials to assign a temporary password that a user must change upon their next login.

**Before you begin**
Configure a Credential Policy, on page 247

**Procedure**

**Step 1**
In Cisco Unified CM Administration, choose User Management > Credential Policy Default.

**Step 2**
From the **Credential Policy** drop-down list box, choose the credential policy for this group.

**Step 3**
Enter the password in both the **Change Credential** and **Confirm Credential** configuration windows.

**Step 4**
Check the **User Cannot Change** check box if you do not want your users to be able to change this credential.

**Step 5**
Check the **User Must Change at Next Login** check box if you want to use this credential as a temporary credential that an end user must change the next time that they login.

**Note**
Please note that, if you check this box, your users are unable to change PIN using Personal Directory service.

**Step 6**
If you do not want the credential to expire, check the **Does Not Expire** check box.

**Step 7**
Click **Save**.
What to do next

If you want to apply a credential policy to a specific end user or PIN:

- Apply Credential Policy to End User, on page 277
Configure User Profiles

• User Profile Overview, on page 249
• User Profile Prerequisites, on page 249
• User Profile Configuration Task Flow, on page 250

User Profile Overview

User profiles contain common directory number and device settings. You can configure different user profiles that contain the most common directory number settings and device settings that your users require and then assign each user profile to the users that require those settings. You can configure different user profiles for different groups of users in your company, according to the phone line and phone setting requirements for each set of users.

For those end users who are enabled for self-provisioning, the phone and phone line settings from the user profile get applied to any new phones that the user provisions. If the user is not enabled for self-provisioning, the user profile settings can be applied to any new phones that the administrator provisions on behalf of the end user.

User profiles use settings from the following phone and phone line templates to build a profile for the end user:

• Universal Line Template—a collection of common phone line settings that are typically assigned to a directory number. Universal line templates allow you to quickly configure phone lines for new directory numbers that get assigned to an end user.

• Universal Device Template—a collection of common device settings that are typically assigned to a phone or other device. Universal device templates allow you to quickly configure new phones that get assigned to an end user.

User Profile Prerequisites

Before configuring your user profiles, make sure to plan how you are going to provision phones for your deployment. Decide whether you are going to use self-provisioning to allow end users to provision their own phones.
# User Profile Configuration Task Flow

## Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Configure a Universal Line Template, on page 250</th>
<th>Purpose: Configure universal line templates with common settings that are typically applied to a directory number.</th>
</tr>
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<tbody>
<tr>
<td>Step 2</td>
<td>Configure a Universal Device Template, on page 251</td>
<td>Purpose: Configure universal device templates with common settings that are typically applied to phones and other devices.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a User Profile, on page 251</td>
<td>Purpose: Assign the universal line and universal device templates to a user profile.</td>
</tr>
</tbody>
</table>

## Configure a Universal Line Template

Universal Line Templates make it easy to apply common settings to newly assigned directory numbers. Configure different templates to meet the needs of different groups of users.

### Procedure

**Step 1**  
In Cisco Unified CM Administration, choose User Management > User/Phone Add > Universal Line Template.

**Step 2**  
Click Add New.

**Step 3**  
Configure the fields in the Universal Line Template Configuration window. See the online help for more information about the fields and their configuration options.

**Step 4**  
If you are deploying Global Dial Plan Replication with alternate numbers expand the Enterprise Alternate Number and +E.164 Alternate Number sections and do the following:

a)  
Click the Add Enterprise Alternate Number button and/or Add +E.164 Alternate Number button.

b)  
Add the Number Mask that you want to use to assign to your alternate numbers. For example, a 4-digit extension might use 5XXXX as an enterprise number mask and 1972555XXXX as an +E.164 alternate number mask.

c)  
Assign the partition where you want to assign alternate numbers.

d)  
If you want to advertise this number via ILS, check the Advertise Globally via ILS check box. Note that if you are using advertised patterns to summarize a range of alternate numbers, you may not need to advertise individual alternate numbers.

e)  
Expand the PSTN Failover section and choose the Enterprise Number or +E.164 Alternate Number as the PSTN failover to use if normal call routing fails.

**Step 5**  
Click Save.
Configure a Universal Device Template

Universal device templates make it easy to apply configuration settings to newly provisioned devices. The provisioned device uses the settings of the universal device template. You can configure different device templates to meet the needs of different groups of users. You can also assign the profiles that you’ve configured to this template.

Procedure

**Step 1**
In Cisco Unified CM Administration, choose User Management > User/Phone Add > Universal Device Template.

**Step 2**
Click Add New.

**Step 3**
Enter the following mandatory fields:
- a) Enter a Device Description for the template.
- b) Select a Device Pool type from the drop-down list.
- c) Select a Device Security Profile from the drop-down list.
- d) Select a SIP Profile from the drop-down list.
- e) Select a Phone Button Template from the drop-down list.

**Step 4**
Complete the remaining fields in the Universal Device Template Configuration window. For field descriptions, see the online help.

**Step 5**
Under Phone Settings, complete the following optional fields:
- a) If you configured a Common Phone Profile, assign the profile.
- b) If you configured a Common Device Configuration, assign the configuration.
- c) If you configured a Feature Control Policy, assign the policy.

**Step 6**
Click Save.

Configure a User Profile

Assign universal line and universal device template to users through the User Profile. Configure multiple user profiles for different groups of users. You can also enable self-provisioning for users who use this service profile.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose User Management > User/Phone Add > User Profile.

**Step 2**
Click Add New.

**Step 3**
Enter a Name and Description for the user profile.

**Step 4**
Assign a Universal Device Template to apply to users' Desk Phones, Mobile and Desktop Devices, and Remote Destination/Device Profiles.

**Step 5**
Assign a Universal Line Template to apply to the phone lines for users in this user profile.

**Step 6**
If you want the users in this user profile to be able to use the self-provisioning feature to provision their own phones, do the following:
a) Check the **Allow End User to Provision their own phones** check box.

b) In the **Limit Provisioning once End User has this many phones** field, enter a maximum number of phones the user is allowed to provision. The maximum is 20.

**Step 7**  
Click Save.
Chapter 35

Configure Service Profile

- Service Profile Overview, on page 253
- Service Profile Configuration Task Flow, on page 254

Service Profile Overview

A Service Profile allows you to create a profile that comprises common Unified Communications (UC) Services settings. You can then apply the service profile to an end user in order to assign the UC services configuration settings in the Service Profile to that end user. You can configure different service profiles for different groups of users in your company so that each group of users has the appropriate services configured for their job.

A Service Profile comprises configuration settings for the following UC services:

- Voicemail
- Mailstore
- Conferencing
- Directory
- IM and Presence
- CTI
- Video conferencing services

Applying Service Profiles to End Users

You can use the following methods to apply a service profile to an end user:

- For LDAP Synchronized Users—If you have imported end users from an LDAP directory, you can assign the service profile to a feature group template and then apply that feature group template to your end users.

- For Active Local Users (i.e. non-LDAP users)—In End User Configuration, you can assign a service profile for an individual end user. You can also use the Bulk Administration Tool to assign a service profile for many end users at once. For details, see the Bulk Administration Guide for Cisco Unified Communications Manager.
Service Profile Configuration Task Flow

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure any of the following Unified Communications (UC) services that you want to assign to this service profile:</td>
</tr>
<tr>
<td></td>
<td>• Add Voicemail Service, on page 254</td>
</tr>
<tr>
<td></td>
<td>• Add Mailstore Service, on page 255</td>
</tr>
<tr>
<td></td>
<td>• Add Conferencing Service, on page 256</td>
</tr>
<tr>
<td></td>
<td>• Add Directory Service, on page 256</td>
</tr>
<tr>
<td></td>
<td>• Add IM and Presence Service, on page 257</td>
</tr>
<tr>
<td></td>
<td>• Add CTI Service, on page 258</td>
</tr>
<tr>
<td></td>
<td>• Add Video Conference Scheduling Service, on page 259</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure the user's service profile to point to the UC services that you want to apply to this service profile.</td>
</tr>
</tbody>
</table>

Add Voicemail Service

Add a voicemail service to your system. You can add multiple voicemail services and then select which service you want to add to your service profiles.

**Procedure**

**Step 1** From Cisco Unified CM Administration choose User Management > User Settings > UC Service.

**Step 2** Click Add New.

**Step 3** From the UC Service Type drop-down list box, choose Voicemail.

**Step 4** From the Product Type drop-down list box, choose Unity or Unity Connection.

**Step 5** Enter a Name for the voicemail service.

**Step 6** Enter a Description that helps you distinguish between services.

**Step 7** In the Hostname/IP Address field, enter the hostname, IP address, or fully qualified domain name of the server that hosts the voicemail service.

**Step 8** In the Port field, enter a port to connect to the voicemail service. The default port is 443.

**Step 9** In the Protocol field, enter the protocol that will be used to route voicemail messages. The available options are HTTP and HTTPS.

**Note** Cisco recommends that you use HTTPS as the voicemail transport protocol for Cisco Unity and Cisco Unity Connection servers. Only change to HTTP if your network configuration does not support HTTPS.
Step 10  
Click Save.

---

What to do next
Add Mailstore Service, on page 255

**Add Mailstore Service**

Add a mailstore service to your system. Cisco Jabber clients use the mailstore service for visual voicemail functionality.

**Note**
Cisco Unity creates subscriber mailboxes for message storage on the Microsoft Exchange server. Cisco Unity Connection usually provides a mailstore service, and hosts the mailstore service on the same server.

**Before you begin**
Add Voicemail Service, on page 254

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose User Management > User Settings > UC Service.

**Step 2**
Click Add New.

**Step 3**
From the UC Service Type drop-down list box, choose Mailstore.

**Step 4**
Enter a Name for the mailstore service.

**Step 5**
Enter a Description for the mailstore service.

**Step 6**
In the Hostname/IP Address field, enter the hostname, IP address, or fully qualified domain name for the server that hosts the mailstore service.

**Step 7**
In the Port field, specify a port between 1–65535 that matches the available port on the mailstore service. The default mailstore port is 143.

**Note**
For secure voice messaging with Cisco Unity connection, use 7993.

**Step 8**
In the Protocol field, enter the protocol that will be used to route voicemail messages: TCP (default), TLS, UDP, or SSL.

**Note**
For secure messaging with Cisco Unity Connection, use TLS.

**Step 9**
Click Save.

---

What to do next
Add Conferencing Service, on page 256
Add Conferencing Service

Add a conferencing service to your system.

Before you begin
Add Mailstore Service, on page 255

Procedure

**Step 1** From Cisco Unified CM Administration, choose **User Management > User Settings > UC Service**.

**Step 2** Click **Add New**.

**Step 3** From the **UC Service Type** drop-down list box, choose **Conferencing**.

**Step 4** From the **Product Type** drop-down list box, choose the product that you want to use for conferencing:

- MeetingPlace Classic
- MeetingPlace Express
- WebEx

**Step 5** Enter a **Name** for the conferencing service.

**Step 6** Enter a **Description** for the conferencing service.

**Step 7** In the **Hostname/IP Address** field, enter the hostname, IP address, or fully qualified domain name of the server that hosts the conferencing service.

**Step 8** In the **Port** field, enter a port value that matches the available port on the conferencing service. The recommended values are:

- 80 (default setting)—Use this port for HTTP
- 443—Use this port for HTTPS

**Step 9** From the **Protocol** drop-down list box, choose the Protocol to use when endpoints contact this service:

- TCP (default setting)
- UDP
- SSL
- TLS

**Note** For secure messaging with Cisco Unity Connection, use TLS.

**Step 10** Click **Save**.

What to do next
Add Directory Service, on page 256

Add Directory Service

Add a directory service to your system if you want to point Cisco Unified Communications Manager towards an external LDAP directory for directory lookups.
Before you begin

Add Conferencing Service, on page 256

Procedure

Step 1
From Cisco Unified CM Administration, choose User Management > User Settings > UC Service.

Step 2
Click Add New.

Step 3
From the UC Service Type drop-down list box, choose Directory.

Step 4
From the Product Type field, choose either of the following:

- Directory—Choose this option if you want your clients to use UDS to connect to the Cisco Unified Communications Manager database for directory lookups.
- Enhanced Directory—Choose this option if you want your clients to connect to an external LDAP directory for directory lookups.

Step 5
Enter a Name for the directory service.

Step 6
Enter a Description for the directory service.

Step 7
In the Hostname/IP Address field, enter the hostname, IP address, or fully qualified domain name of the server that hosts the directory service that you want your clients to use for directory lookups.

Note If you are using an external LDAP directory for directory lookups, enter the hostname, IP address, or fully qualified domain name of the LDAP directory.

Step 8
In the Port field, enter a port number that matches the available port on the directory service. The default port value is 389. In addition, ports 636, 3628, 3629 can connect to an external LDAP directory.

Step 9
In the Protocol field, enter the protocol that will be used to route communications between the directory service and endpoints. The available options are:

- TCP (default setting)
- UDP
- TLS

Step 10
Click Save.

What to do next

Add IM and Presence Service, on page 257

Add IM and Presence Service

Add an IM and Presence service to your system.

Before you begin

Add Directory Service, on page 256
**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **User Management > User Settings > UC Service**.

**Step 2**  
Click **Add New**.

**Step 3**  
From the **UC Service Type** drop-down list box, choose IM and Presence.

**Step 4**  
From the **Product Type** drop-down list box, choose one of the following options:
- Unified CM (IM and Presence)
- WeEx (IM and Presence)

**Step 5**  
Enter a **Name** for the IM and Presence service.

**Step 6**  
Enter a **Description** for the IM and Presence service.

**Step 7**  
In the **Hostname/IP Address** field, enter the hostname, IP address, or DNS SRV for the server that hosts the IM and Presence service.

**Tip**  
Cisco recommends DNS SRV to help the client find the correct IM and Presence service for the user.

**Step 8**  
Click **Save**.

---

**What to do next**

*Add CTI Service, on page 258*

---

**Add CTI Service**

Add a CTI service to your system.

**Before you begin**

*Add IM and Presence Service, on page 257*

---

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **User Management > User Settings > UC Service**.

**Step 2**  
Click **Add New**.

**Step 3**  
From the **UC Service Type** drop-down list box, choose **CTI**.

**Step 4**  
Enter a **Name** for the CTI service.

**Step 5**  
Enter a **Description** for the CTI service.

**Step 6**  
In the **Hostname/IP Address** field, enter the hostname, IP address, or fully qualified domain name for the server that hosts the CTI service.

**Step 7**  
In the **Port** field, enter the port number of the CTI service. The default port is 2748.

**Step 8**  
Click **Save**.
Add Video Conference Scheduling Service

Add a video conference scheduling service that provides a portal to the TelePresence Management System for video conference scheduling.

**Before you begin**

Add CTI Service, on page 258

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose User Management > User Settings > UC Service.

**Step 2**
Click Add New.

**Step 3**
Enter a Name for the service.

**Step 4**
Enter a Description for the service.

**Step 5**
In the IP Address/Hostname field, enter the hostname, IP address, or fully qualified domain name of the server that hosts the video conferencing scheduling service.

**Step 6**
In the Port field, enter a port number that matches the available port on the video conference scheduling service. The available ports are:

- 80 (default) or 8080—use these ports for HTTP
- 443 or 8443—use these ports for HTTPS

**Step 7**
From the Protocol drop-down list box, choose one of the following protocols for communications with the video conference scheduling service:

- HTTP
- HTTPS

**Step 8**
In the Portal URL field, enter a URL that points to the TelePresence Management System.

**Step 9**
Click Save.

**What to do next**

Configure a Service Profile, on page 259

**Configure a Service Profile**

Configure a Service Profile that include the UC Services that you want to assign to end users who use the profile.

**Before you begin**

You must set up your Unified Communications (UC) services before you can add them to a service profile.
### Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>User Management &gt; User Settings &gt; Service Profile</strong>.</th>
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</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Add New</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Enter a <strong>Name</strong> for the chosen Service Profile Configuration.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Enter a <strong>Description</strong> for the chosen Service Profile Configuration.</td>
</tr>
<tr>
<td>Step 5</td>
<td>For each UC service that you want to be a part of this profile, assign the <strong>Primary</strong>, <strong>Secondary</strong>, and <strong>Tertiary</strong> connections for that service.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Complete the remaining fields in the <strong>Service Profile Configuration</strong> window. For detailed field descriptions, see the online help.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>
Configure Feature Group Template

• Feature Group Template Overview, on page 261
• Feature Group Template Prerequisites, on page 261
• Configure a Feature Group Template, on page 262

Feature Group Template Overview

Feature group templates help you deploy your end users with configured phones and phone lines. Feature group templates allow you to assign common phone, phone line, and service settings to all users who are assigned that feature group template. If you have also enabled self-provisioning for your end users, the feature group template allows your users to quickly provision and set up their phones with the desired phone, phone line and services settings.

The feature group template configuration includes the following profiles that you can assign to the feature group template:

• User Profile—contains a set of common phone and phone line settings. You must configure the user profile with a universal line template, which assigns the common phone line settings, and a universal device template, which assigns the common phone settings. These templates assist users who are set up for self-provisioning to configure their own phones.

• Service Profile—contains a group of common settings for Unified Communications services such as conferencing and directory services.

When you configure a feature group template to include a user profile and service profile and then assign that feature group template to an end user, the user profile and service profile settings propagate through to any new phones that the end user provisions.

If you are deploying the IM and Presence Service, you can use the feature group template to enable LDAP-synchronized users with instant messaging and presence capability.

Feature Group Template Prerequisites

Before you configure a feature group template, configure a user profile and service profile for your end users.

• User Profile Configuration Task Flow, on page 250
• Service Profile Configuration Task Flow, on page 254
Configure a Feature Group Template

Feature group templates aid in your system deployment by helping you to quickly configure phones, lines, and features for your provisioned users. If you are syncing users from a company LDAP directory, configure a feature group template with the User Profile and Service Profile that you want users synced from the directory to use. You can also enable the IM and Presence Service for synced users through this template.

**Procedure**

**Step 1**  
In Cisco Unified CM Administration, choose **User Management > User/Phone Add > Feature Group Template**.

**Step 2**  
Click **Add New**.

**Step 3**  
Enter a **Name** and **Description** for the Feature Group Template.

**Step 4**  
Check the **Home Cluster** check box if you want to use the local cluster as the home cluster for all users whom use this template.

**Step 5**  
Check the **Enable User for Unified CM IM and Presence** check box to allow users whom use this template to exchange instant messaging and presence information.

**Step 6**  
From the drop-down list, select a **Services Profile** and **User Profile**.

**Step 7**  
Complete the remaining fields in the **Feature Group Template Configuration** window. Refer to the online help for field descriptions.

**Step 8**  
Click **Save**.

**What to do next**

Associate the feature group template with an LDAP directory sync to apply the settings from the template to synchronized end users.
CHAPTER 37

Import Users From LDAP Directory

- LDAP Synchronization Overview, on page 263
- LDAP Synchronization Prerequisites, on page 265
- LDAP Synchronization Configuration Task Flow, on page 265

LDAP Synchronization Overview

Lightweight Directory Access Protocol (LDAP) synchronization helps you to provision and configure end users for your system. During LDAP synchronization, the system imports a list of users and associated user data from an external LDAP directory into the Unified Communications Manager database. You can also configure your end users while the import occurs.

Note
Unified Communications Manager supports LDAPS (LDAP with SSL) but does not support LDAP with StartTLS. Ensure that you upload the LDAP server certificate to Unified Communications Manager as a Tomcat-Trust.

See the Compatibility Matrix for Cisco Unified Communications Manager and the IM and Presence Service for information on the supported LDAP directories.

LDAP synchronization advertises the following functionalities:

- **Importing End Users**—You can use LDAP synchronization during the initial system setup to import your user list from a company LDAP directory into the Unified Communications Manager database. If you've preconfigured items such as feature group templates, user profiles, service profiles, universal device and line templates, you can apply configurations to your users, and assign configured directory numbers and directory URIs during the sync process. The LDAP synchronization process imports the list of users and user-specific data and applies the configuration templates that you've set up.

  Note
  You cannot make edits to an LDAP synchronization once the initial synchronization has occurred already.

- **Scheduled Updates**—You can configure Unified Communications Manager to synchronize with multiple LDAP directories at scheduled intervals to ensure that the database is updated regularly and user data is up-to-date.
• **Authenticate End Users**—You can configure your system to authenticate end user passwords against the LDAP directory rather than the Cisco Unified Communications Manager database. LDAP authentication provides companies with the ability to assign a single password to end users for all company applications. This functionality does not apply to PINs or application user passwords.

• **Directory Server User Search for Cisco MRA Clients and Endpoints**—You can search a corporate directory server even when operating outside the enterprise firewall. When this feature is enabled, the User Data Service (UDS) acts as a proxy and sends the user search request to the corporate directory instead of sending it to the Unified Communications Manager database.

---

**LDAP Authentication for End Users**

LDAP synchronization allows you to configure your system to authenticate end user passwords against the LDAP directory rather than the Cisco Unified Communications Manager database. LDAP authentication provides companies with the ability to assign a single password to end users for all company applications. This functionality does not apply to PINs or application user passwords.

**Directory Server User Search for Cisco Mobile and Remote Access Clients and Endpoints**

In previous releases, when a user with a Cisco mobile and remote access client (for example, Cisco Jabber) or endpoint (for example, Cisco DX 80 phone) performed a user search while outside the enterprise firewall, results were based on those user accounts that are saved in the Cisco Unified Communications Manager database. The database contains user accounts which are either configured locally or synchronized from the corporate directory.

With this release, Cisco mobile and remote access clients and endpoints can now search a corporate directory server even when operating outside the enterprise firewall. When this feature is enabled, the User Data Service (UDS) acts as a proxy and sends the user search request to the corporate directory instead of sending it to the Cisco Unified Communications Manager database.

Use this feature to achieve the following results:

- Deliver the same user search results regardless of geographic location—Mobile and remote access clients and endpoints can perform user searches by using the corporate directory; even when they are connected outside the enterprise firewall.

- Reduce the number of user accounts that are configured in the Cisco Unified Communications Manager database—Mobile clients can now search users in the corporate directory. In the previous releases, user search results were based on the users that are configured in the database. Now, administrators no longer need to configure or synchronize user accounts to the database solely for user searches. Administrators need to configure only those user accounts that are served by a cluster. Reducing the total number of user accounts in the database shortens software upgrade time frames while improving overall database performance.

To configure this feature, you must enable the **Enable user search to Enterprise Directory Server** option in the **LDAP Search Configuration** window, and configure the LDAP directory server details. For details, see the **Configure Enterprise Directory User Search, on page 269** procedure.
LDAP Synchronization Prerequisites

Prerequisite Tasks

Before you import end users from an LDAP directory, complete the following tasks:

- **Configure User Access.** Decide which access control groups you want to assign to your users. For many deployments, the default groups are sufficient. If you need to customize your roles and groups, refer to the 'Manage User Access' chapter of the Administration Guide.

- **Configure Default credentials for a credential policy that is applied by default to newly provisioned users.**

- **If you are syncing users from an LDAP directory, make sure that you have a Feature Group Template set up that includes the User Profiles, Service Profiles, and Universal Line and Device Template settings that you want to assign to your users phones and phone extensions.**

---

**Note**

For users whose data you want to synchronize to your system, ensure that their email ID fields on the Active Directory server are unique entries or left blank.

LDAP Synchronization Configuration Task Flow

Use the following tasks to pull a user list from the external LDAP directory and import it into the Unified Communications Manager database.

---

**Note**

If you have already synced the LDAP directory once, you can still sync new items from your external LDAP directory, but you cannot add new configurations in Unified Communications Manager to the LDAP directory sync. In this case, you can use the Bulk Administration Tool and menus such as Update Users or Insert Users. Refer to the Bulk Administration Guide for Cisco Unified Communications Manager.

---

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> <a href="#">Activate the Cisco DirSync Service, on page 266</a></td>
<td>Log in to Cisco Unified Serviceability and activate the Cisco DirSync service.</td>
</tr>
<tr>
<td><strong>Step 2</strong> <a href="#">Enable LDAP Directory Synchronization, on page 266</a></td>
<td>Enable LDAP directory synchronization in Unified Communications Manager.</td>
</tr>
<tr>
<td><strong>Step 3</strong> <a href="#">Create an LDAP Filter, on page 267</a></td>
<td><strong>Optional.</strong> Create an LDAP filter if you want Unified Communications Manager to synchronize only a subset of users from your corporate LDAP directory.</td>
</tr>
</tbody>
</table>
Activate the Cisco DirSync Service

Perform this procedure to activate the Cisco DirSync Service in Cisco Unified Serviceability. You must activate this service if you want to synchronize end user settings from a corporate LDAP directory.

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified Serviceability, choose <strong>Tools &gt; Service Activation</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>From the <strong>Server</strong> drop-down list, choose the publisher node.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Under <strong>Directory Services</strong>, click the <strong>Cisco DirSync</strong> radio button.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
<td></td>
</tr>
</tbody>
</table>

Enable LDAP Directory Synchronization

Perform this procedure if you want to configure Unified Communications Manager to synchronize end user settings from a corporate LDAP directory.

**Note**

If you have already synced the LDAP directory once, you can still sync new users from your external LDAP directory, but you cannot add new configurations in Unified Communications Manager to the LDAP directory sync. You also cannot add edits to underlying configuration items such as the feature group template or user profile. If you have already completed one LDAP sync, and want to add users with different settings, you can use Bulk Administration menus such as Update Users or Insert Users.
Procedure

Step 1 From Cisco Unified CM Administration, choose System > LDAP > LDAP System.
Step 2 If you want Unified Communications Manager to import users from your LDAP directory, check the Enable Synchronizing from LDAP Server check box.
Step 3 From the LDAP Server Type drop-down list, choose the type of LDAP directory server that your company uses.
Step 4 From the LDAP Attribute for User ID drop-down list, choose the attribute from your corporate LDAP directory that you want Unified Communications Manager to synchronize with for the User ID field in the End User Configuration window.
Step 5 Click Save.

Create an LDAP Filter

You can create an LDAP filter to limit your LDAP synchronization to a subset of users from your LDAP directory. When you apply the LDAP filter to your LDAP directory, Unified Communications Manager imports only those users from the LDAP directory who match the filter.

Note

Any LDAP filter that you configure must comply with the LDAP search filter standards that are specified in RFC4515.

Procedure

Step 1 In Cisco Unified CM Administration, choose System > LDAP > LDAP Filter.
Step 2 Click Add New to create a new LDAP filter.
Step 3 In the Filter Name text box, enter a name for your LDAP filter.
Step 4 In the Filter text box, enter a filter. The filter can contain a maximum of 1024 UTF-8 characters and must be enclosed in parentheses ( ).
Step 5 Click Save.

Configure LDAP Directory Sync

Use this procedure to configure Unified Communications Manager to synchronize with an LDAP directory. LDAP directory synchronization allows you to import end user data from an external LDAP directory into the Unified Communications Manager database such that it displays in End User Configuration window. If you have setup feature group templates with universal line and device templates, you can assign settings to newly provisioned users and their extensions automatically.
If you are assigning access control groups or feature group templates, you can use an LDAP filter to limit the import to the group of users with the same configuration requirements.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **System > LDAP > LDAP Directory**.

**Step 2**
Perform one of the following steps:
- Click **Find** and select an existing LDAP directory.
- Click **Add New** to create a new LDAP directory.

**Step 3**
In the **LDAP Directory Configuration** window, enter the following:

- In the **LDAP Configuration Name** field, assign a unique name to the LDAP directory.
- In the **LDAP Manager Distinguished Name** field, enter a userID with access to the LDAP directory server.
- Enter and confirm the password details.
- In the **LDAP User Search Space** field, enter the search space details.
- In the **LDAP Custom Filter for Users Synchronize** field, select either **Users Only** or **Users and Groups**.
- (Optional). If you want to limit the import to only a subset of users who meet a specific profile, from the **LDAP Custom Filter for Groups** drop-down list, select an LDAP filter.

**Step 4**
In the **LDAP Directory Synchronization Schedule** fields, create a schedule that Unified Communications Manager uses to synchronize data with the external LDAP directory.

**Step 5**
Complete the **Standard User Fields to be Synchronized** section. For each End User field, choose an LDAP attribute. The synchronization process assigns the value of the LDAP attribute to the end user field in Unified Communications Manager.

**Step 6**
If you are deploying URI dialing, make sure to assign the LDAP attribute that will be used for the user's primary directory URI address.

**Step 7**
In the **Custom User Fields To Be Synchronized** section, enter custom user field name with the required LDAP attribute.

**Step 8**
To assign the imported end users to an access control group that is common to all the imported end users, do the following:
- Click **Add to Access Control Group**.
- In the pop-up window, click the corresponding check box for each access control group that you want to assign to the imported end users.
- Click **Add Selected**.

**Step 9**
If you want to assign a feature group template, select the template from the **Feature Group Template** drop-down list.

**Note**
The end users are synced with the assigned **Feature Group Template** only for the first time when the users are not present. If an existing **Feature Group Template** is modified and a full sync is performed for the associated LDAP, the modifications will not get updated.

**Step 10**
If you want to assign primary extension by applying a mask to imported telephone numbers, do the following:
- Check the **Apply mask to synced telephone numbers to create a new line for inserted users** check box.
b) Enter a **Mask**. For example, a mask of 11XX creates a primary extension of 1145 if the imported telephone number is 8889945.

**Step 11**
If you want to assign primary extensions from a pool of directory numbers, do the following:

a) Check the **Assign new line from the pool list if one was not created based on a synced LDAP telephone number** check box.

b) In the **DN Pool Start** and **DN Pool End** text boxes, enter the range of directory numbers from which to select primary extensions.

**Step 12**
In the **LDAP Server Information** section, enter the hostname or IP address of the LDAP server.

**Step 13**
If you want to use TLS to create a secure connection to the LDAP server, check the **Use TLS** check box.

**Step 14**
Click **Save**.

**Step 15**
To complete an LDAP sync, click **Perform Full Sync Now**. Otherwise, you can wait for the scheduled sync.

---

**Note**
When users are deleted in LDAP, they will automatically be removed from Unified Communications Manager after 24 hours. Also, if the deleted user is configured as a mobility user for any of the following devices, these inactive devices will also be automatically deleted:

- Remote Destination Profile
- Remote Destination Profile Template
- Mobile Smart Client
- CTI Remote Device
- Spark Remote Device
- Nokia S60
- Cisco Dual Mode for iPhone
- IMS-integrated Mobile (Basic)
- Carrier-integrated Mobile
- Cisco Dual Mode for Android

---

**Configure Enterprise Directory User Search**

Use this procedure to configure phones and clients in your system to perform user searches against an enterprise directory server instead of the database.

**Before you begin**

- Ensure that the primary, secondary, and tertiary servers, which you choose for LDAP user search, are network reachable to the Unified Communications Manager subscriber nodes.

- From **System > LDAP > LDAP System**, configure the type of LDAP server from the **LDAP Server Type** drop-down list in the **LDAP System Configuration** window.
Procedure

Step 1  In Cisco Unified CM Administration, choose System > LDAP > LDAP Search.
Step 2  To enable user searches to be performed using an enterprise LDAP directory server, check the Enable user search to Enterprise Directory Server check box.
Step 3  Configure the fields in the LDAP Search Configuration window. See the online help for more information about the fields and their configuration options.
Step 4  Click Save.

LDAP Attributes for UDS Search of Directory Server

The following table lists the LDAP attributes that UDS users search request uses when the Enable user search to Enterprise Directory Server option is enabled. For these types of directory requests, UDS acts as a proxy and relays the search request to the corporate directory server.

<table>
<thead>
<tr>
<th>UDS Users Response Tag</th>
<th>LDAP Attribute</th>
</tr>
</thead>
<tbody>
<tr>
<td>userName</td>
<td>• samAccountName</td>
</tr>
<tr>
<td></td>
<td>• uid</td>
</tr>
<tr>
<td>firstName</td>
<td>givenName</td>
</tr>
<tr>
<td>lastName</td>
<td>sn</td>
</tr>
<tr>
<td>middleName</td>
<td>• initials</td>
</tr>
<tr>
<td></td>
<td>• middleName</td>
</tr>
<tr>
<td>nickName</td>
<td>nickName</td>
</tr>
<tr>
<td>displayName</td>
<td>displayName</td>
</tr>
<tr>
<td>phoneNumber</td>
<td>• telephonenumber</td>
</tr>
<tr>
<td></td>
<td>• ipPhone</td>
</tr>
<tr>
<td>homeNumber</td>
<td>homephone</td>
</tr>
<tr>
<td>mobileNumber</td>
<td>mobile</td>
</tr>
<tr>
<td>email</td>
<td>mail</td>
</tr>
</tbody>
</table>

Note  UDS users response tag may be mapped to one of the LDAP attributes. The mapping of the attributes is determined by the option you select from the LDAP Server Type drop-down list. Access this drop-down list from System > LDAP > LDAP System Configuration window.
Configure LDAP Authentication

Perform this procedure if you want to enable LDAP authentication so that end user passwords are authenticated against the password that is assigned in the company LDAP directory. This configuration applies to end user passwords only and does not apply to end user PINs or application user passwords.

**Procedure**

**Step 1** In Cisco Unified CM Administration, choose **System > LDAP > LDAP Authentication**.

**Step 2** Check the **Use LDAP Authentication for End Users** check box to use your LDAP directory for user authentication.

**Step 3** In the **LDAP Manager Distinguished Name** field, enter the user ID of the LDAP Manager who has access rights to the LDAP directory.

**Step 4** In the **Confirm Password** field, enter the password for the LDAP manager.

**Step 5** In the **LDAP User Search Base** field, enter the search criteria.

**Step 6** In the **LDAP Server Information** section, enter the hostname or IP address of the LDAP server.

**Step 7** If you want to use TLS to create a secure connection to the LDAP server, check the **Use TLS** check box.

**Step 8** Click **Save**.

**What to do next**

Customize LDAP Agreement Service Parameters, on page 271

### Customize LDAP Agreement Service Parameters

Perform this procedure to configure the optional service parameters that customize the system-level settings for LDAP agreements. If you do not configure these service parameters, Unified Communications Manager applies the default settings for LDAP directory integration. For parameter descriptions, click the parameter name in the user interface.

<table>
<thead>
<tr>
<th>UDS Users Response Tag</th>
<th>LDAP Attribute</th>
</tr>
</thead>
<tbody>
<tr>
<td>directoryUri</td>
<td>• msRTCSIP-primaryuseraddress</td>
</tr>
<tr>
<td></td>
<td>• mail</td>
</tr>
<tr>
<td>department</td>
<td>• department</td>
</tr>
<tr>
<td></td>
<td>• departmentNumber</td>
</tr>
<tr>
<td>manager</td>
<td>manager</td>
</tr>
<tr>
<td>title</td>
<td>title</td>
</tr>
<tr>
<td>pager</td>
<td>pager</td>
</tr>
</tbody>
</table>
You can use service parameters to customize the below settings:

- **Maximum Number of Agreements**—Default value is 20.
- **Maximum Number of Hosts**—Default value is 3.
- **Retry Delay On Host Failure (secs)**—Default value for host failure is 5.
- **Retry Delay On HotList failure (mins)**—Default value for hostlist failure is 10.
- **LDAP Connection Timeouts (secs)**—Default value is 5.
- **Delayed Sync Start time (mins)**—Default value is 5.
- **User Customer Map Audit Time**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **System > Service Parameters**.

**Step 2** From the **Server** drop-down list box, choose the publisher node.

**Step 3** From the **Service** drop-down list box, choose **Cisco DirSync**.

**Step 4** Configure values for the Cisco DirSync service parameters.

**Step 5** Click **Save**.

**LDAP Directory Service Parameters**

<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Number Of Agreements</td>
<td>The maximum number of LDAP directories that you can configure. The default setting is 20.</td>
</tr>
<tr>
<td>Maximum Number Of Hosts</td>
<td>The maximum number of LDAP hostnames that you can configure for failover purposes. The default value is 3.</td>
</tr>
<tr>
<td>Retry Delay On Host Failure (secs)</td>
<td>After a host failure, the number of seconds that Cisco Unified Communications Manager delays before it retries the connection to the first LDAP server (hostname). The default value is 5.</td>
</tr>
<tr>
<td>Retry Delay On HotList Failure (mins)</td>
<td>After a hostlist failure, the number of minutes that Cisco Unified Communications Manager delays before it retries every configured LDAP server (hostnames). The default is 10.</td>
</tr>
<tr>
<td>LDAP Connection Timeout (secs)</td>
<td>The number of seconds that Cisco Unified Communications Manager allows for establishing the LDAP connection. The LDAP service provider aborts the connection attempt if a connection cannot be established in the specified amount of time. The default is 5.</td>
</tr>
<tr>
<td>Delayed Sync Start time (mins)</td>
<td>The number of minutes that Cisco Unified Communications Manager delays in starting the directory synchronization process after the Cisco DirSync service starts. The default is 5.</td>
</tr>
</tbody>
</table>
Convert LDAP Synchronized User to Local User

When you synchronize your LDAP directory with Cisco Unified Communications Manager, for LDAP-synchronized end users, you cannot edit any of the fields within the End User Configuration window unless you convert the LDAP-synchronized user to a local user.

To edit to an LDAP-synchronized field in the End User Configuration window, convert the user to a local user. However, if you perform this conversion, the end user will not be updated when Cisco Unified Communications Manager synchronizes with the LDAP directory.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified CM Administration, choose End Users &gt; End User Management.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select the end user.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click the Convert to Local User button.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Make your updates in the End User Configuration window.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Assign LDAP Synchronized Users to an Access Control Group

Perform this procedure to assign LDAP synchronized users to an access control group.

Before you begin

Cisco Unified Communications Manager must be configured to synchronize end users with an external LDAP directory.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified CM Administration, choose System &gt; LDAP &gt; LDAP Directory.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select a configured LDAP Directory.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click the Add to Access Control Group button.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Select the access control groups that you want to apply to the end users in this LDAP directory.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Add Selected.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click Perform Full Sync.</td>
</tr>
</tbody>
</table>

Cisco Unified Communications Manager syncs with the external LDAP directory and synchronized users get inserted into the correct access control group.

Note: The synchronized users get inserted into the selected access group only when you add an access control group for the first time. Any subsequent group that you add to LDAP will not be applied to the synchronized users after performing a full sync.
Assign LDAP Synchronized Users to an Access Control Group
Provision End Users Manually

- Manual End User Provision Overview, on page 275
- Manual End User Provisioning Prerequisites, on page 275
- Import End Users using Bulk Administration, on page 275
- Manual End User Configuration Task Flow, on page 276

Manual End User Provision Overview

If you are not importing end users from an LDAP directory, you can use either of the following methods to add your end users to the Unified Communications Manager database:

- Import using Bulk Administration Tool
- Manually add new users

Manual End User Provisioning Prerequisites

Before importing your end users, plan and configure the roles, access control groups, and credential policies for your end users.

- User Access Configuration Task Flow, on page 230
- Credential Policy Configuration Task Flow, on page 246

Import End Users using Bulk Administration

Using the Bulk Administration Tool, you can perform bulk transactions to the Cisco Unified Communications Manager database, including importing and updating large numbers of end users, phones, and ports, in a single process. The Bulk Administration Tool allows you to import the end user list, and end user configurations, from a CSV file into the database.

For details on how to use the Bulk Administration Tool to import end users, see the Bulk Administration Guide for Cisco Unified Communications Manager.
Manual End User Configuration Task Flow

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Add a New End User, on page 276</td>
<td>Add a new end user to the database manually.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Assign End Users to an Access Control Group, on page 277</td>
<td>Assign any local end users that you provision to an access control group with the required role permissions. Local users include any manually provisioned end users and any end users whom you import using the Bulk Administration Tool. Local users have a User Status of 'Active Local User' in <strong>End User Configuration</strong>.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Apply Credential Policy to End User, on page 277</td>
<td>(Optional). Confirm if the default credential policy can be applied to this end user. If not, apply a credential policy to the end user PIN or password.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Assign Feature Group Template to Local End Users, on page 278</td>
<td>Assign a feature group template for the end user. When you assign the feature group template, the system assigns to the end user the user profile, service profile, universal line and device templates, and self-provisioning settings that are associated to that feature group template.</td>
</tr>
</tbody>
</table>

### Add a New End User

Use this procedure to add a new end user to the Unified Communications Manager database manually.

**Procedure**

- **Step 1** In Cisco Unified CM Administration, choose **User Management > End User**.
- **Step 2** Click **Add New**.
- **Step 3** Enter the **User ID** and **Last name**.
- **Step 4** Choose the **User Rank** from the drop-down list.
- **Step 5** Complete the fields in the **End User Configuration** window. For field descriptions, see the online help.
- **Step 6** Click **Save**.

**What to do next**

Assign End Users to an Access Control Group, on page 277
Assign End Users to an Access Control Group

Use this procedure to assign provisioned users to an access control group. For LDAP Synchronized users that were assigned to an access control group during the LDAP sync, you can also use this procedure to assign additional access control groups. This may be useful if your LDAP sync configuration included a common access control group, but you require that some users be assigned to additional access control groups based on their role.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose User Management &gt; User Settings &gt; Access Control Group.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select an access control group.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Add End Users to Group.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Find and List Users window, select the end users whom you want to add to the group.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Add Selected.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Apply Credential Policy to End User

Apply a configured credential policy to a specific end user password or end user PIN. You may need to do this if you need to make an update from the default credential policy.

**Note**

You can also apply a credential policy to an application user password. For details, see the Administration Guide for Cisco Unified Communications Manager.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose User Management &gt; End User.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select the end user.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click the Edit Credential button that corresponds to the password or PIN, depending on the credential to which you want to apply the credential policy.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Authentication Rule drop-down list, choose the credential policy that you want to apply.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Complete any additional fields in the Credential Configuration window. For help with the fields and their settings, see the online help.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>
Assign Feature Group Template to Local End Users

Assign a feature group template to a local end user. A local end user is an end user who has been either added to the database manually, or imported using the Bulk Administration Tool. Local end users are not synchronized with an external LDAP directory.

Procedure

**Step 1**
In Cisco Unified CM Administration, choose **User Management > User/Phone Add > Quick User/Phone Add**.

**Step 2**
Click **Find** and select an end user.

**Step 3**
From the **Feature Group Template** drop-down list, select the feature group template that you have configured for this end user.

**Step 4**
Click **Save**.
PART VI

Configure Endpoint Devices

- Endpoint Devices Overview, on page 281
- Configure Analog Telephone Adaptors, on page 283
- Configure Software-Based Endpoints, on page 311
- Configure Cisco IP Phones, on page 321
- Configure Diagnostics and Reporting for Cisco Unified IP Phones, on page 345
- Configure Third-Party SIP Phones, on page 359
- Device Profiles and Templates, on page 365
- Associate Users with Endpoints, on page 377
CHAPTER 39

Endpoint Devices Overview

• About Endpoint Device Configuration, on page 281
• Endpoint Device Configuration, on page 281

About Endpoint Device Configuration

The chapters in this part provide information about how to configure endpoint devices, and how to associate users with endpoints.

Endpoint Device Configuration

Complete the following task flows to configure end users for your system.

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Analog Telephone Adaptor, on page 284</td>
<td>Configure an analog telephone adapter that acts as an interface between analog telephones and IP-based telephony networks.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Software-Based Endpoint Configuration, on page 311</td>
<td>Configure software-based endpoints such as CTI ports, H.323 clients, and Cisco IP Communicator.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Cisco IP Phones Configuration Task Flow, on page 321</td>
<td>Configure Cisco IP Phones to function on your network.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Diagnostics and Reporting Configuration Task Flow, on page 347</td>
<td>Use call diagnostics and the Quality Reporting Tool (QRT) to ensure call quality on Cisco IP Phones.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Third-Party SIP Endpoints Configuration Task Flow, on page 359</td>
<td>Configure third-party SIP endpoints.</td>
</tr>
</tbody>
</table>
### Endpoint Device Configuration

<table>
<thead>
<tr>
<th>Step 6</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Configure Device Profiles and Templates Task Flow, on page 366</td>
<td>Configure the profiles and templates that define the services, features, and directory numbers that associate with a particular device.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Users and Devices Configuration Task Flow, on page 377</td>
<td>Associate devices with end users and application users.</td>
</tr>
</tbody>
</table>
Configure Analog Telephone Adaptors

• Analog Telephone Adaptor Overview, on page 283
• Configure Analog Telephone Adaptor, on page 284

Analog Telephone Adaptor Overview

The Cisco Analog Telephone Adaptor (ATA) functions as an analog telephone adapter that interfaces regular analog telephones to IP-based telephony networks. The Cisco ATA converts any regular analog telephone into an Internet telephone. Each adapter supports two voice ports, each with its own telephone number.

Like other IP devices, the Cisco ATA receives its configuration file and list of Unified Communications Managers from the TFTP server. If the TFTP server does not have a configuration file, the Cisco ATA uses the TFTP server name or IP address and port number as the primary Unified Communications Manager name or IP address and port number.

The Cisco ATA:

• Contains a single 10 BaseT RJ-45 port and two RJ-11 FXS standard analog telephone ports
• Supports a number of codecs, including G.711 alaw, G.711 ulaw, and G.723 and G.729a voice codecs
• Converts voice into IP data packets
• Supports redial, speed dial, call forwarding, call waiting, call hold, transfer, conference, voice messaging, message-waiting indication, off-hook ringing, caller-ID, callee-ID, and call waiting caller-ID

The ATA 180 series uses SCCP, while the ATA 190 series uses SIP. For more information, see the ATA documentation:

Configure Analog Telephone Adaptor

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.
Step 2 Click Add New.
Step 3 From the Phone Type drop down list, select the Analog Telephone Adaptor model you have and click Next. The Phone Configuration window appears.
Step 4 Configure the fields in the Phone Configuration window.
Step 5 Click Save.
Step 6 Click Apply Config for your changes to take effect and synchronize the phone.

Related Topics
Analog Telephone Adaptor 186 Configuration Fields, on page 284
Analog Telephone Adaptor 187 Configuration Fields, on page 289
Analog Telephony Adaptor 190 Configuration Fields, on page 299

Analog Telephone Adaptor 186 Configuration Fields

Table 24: Analog Telephony Adaptor 186 Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| MAC Address | Enter the Media Access Control (MAC) address that identifies ATA 186. Make sure that the value comprises 12 hexadecimal characters. You can determine the MAC address for ATA 186 in any of these ways:  
  • Look at the MAC label on the back of ATA 186.  
  • Display the web page for ATA 186 and click the Device Information hyperlink. |
| Description | Enter a text description of the ATA 186. This field can contain up to 128 characters. You can use all characters except quotes ("), close angle bracket (>) open angle bracket (<), backslash (\), ampersand (&), and percent sign (%). |
| Device Pool | Choose the device pool to which you want the ATA 186 assigned. The device pool defines sets of common characteristics for devices, such as region, date/time group, and softkey template.  
  To see the Device Pool configuration settings, click the View Details link. |
<table>
<thead>
<tr>
<th><strong>Field</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Common Device Configuration</td>
<td>Choose the Common Device Configuration to which you want the ATA 186 assigned. To see the Common Device Configuration settings, click the View Details link.</td>
</tr>
<tr>
<td>Phone Button Template</td>
<td>Choose the appropriate phone button template. The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.</td>
</tr>
<tr>
<td>Common Phone Profile</td>
<td>From the drop-down list, choose a common phone profile from the list of available common phone profiles. To see the Common Phone Profile settings, click the View Details link.</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>From the drop-down list, choose the calling search space or leave the calling search space as the default of &lt;None&gt;.</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>From the drop-down list, choose the appropriate calling search space for the device to use when it performs automated alternate routing (AAR) or leave the calling search space as the default of &lt;None&gt;.</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Choose the appropriate Media Resource Group List. A Media Resource Group List comprises a prioritized grouping of media resource groups. If you choose &lt;None&gt;, Cisco Unified CM uses the Media Resource Group List that is defined in the device pool.</td>
</tr>
<tr>
<td>Location</td>
<td>From the drop-down list, choose the location that is associated with the phones and gateways in the device pool.</td>
</tr>
<tr>
<td>AAR Group</td>
<td>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. Cisco Unified CM uses the AAR group that is associated with Device Pool or Line.</td>
</tr>
<tr>
<td>User Locale</td>
<td>From the drop-down list, choose the user locale that is associated with the ATA 186. The user locale identifies a set of detailed information to support users, including language and font. If you do not specify a user locale, Cisco Unified CM uses the user locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Network Locale</td>
<td>From the drop-down list, choose the network locale that is associated with the ATA 186. The network locale contains a definition of the tones and cadences that the phone in a specific geographic area uses. If you do not specify a network locale, Cisco Unified CM uses the network locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Device Mobility Mode</td>
<td>From the drop-down list, turn the device mobility feature on or off for this device or choose Default to use the default device mobility mode. Default setting uses the value for the Device Mobility Mode service parameter for a device.</td>
</tr>
<tr>
<td>Owner</td>
<td>Select <strong>User</strong> or <strong>Anonymous</strong> (Public/Shared Space, for the owner type.</td>
</tr>
</tbody>
</table>

**Configure Endpoint Devices**

**Analog Telephone Adaptor 186 Configuration Fields**

---

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<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Owner User ID</td>
<td>From the drop-down list, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device. Assigning a user ID to the device also moves the device from “Unassigned Devices” to “Users” in the License Usage Report.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> Do not configure this field if you are using extension mobility. Extension mobility does not support device owners.</td>
</tr>
<tr>
<td>Phone Load Name</td>
<td>Enter the custom software for ATA 186.</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Choose one of the following values:</td>
</tr>
<tr>
<td></td>
<td>• Off—Choose this value to disable the use of a Trusted Relay Point (TRP) with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates.</td>
</tr>
<tr>
<td></td>
<td>• On—Choose this value to enable the use of a TRP with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates.</td>
</tr>
<tr>
<td></td>
<td>• Default—If you choose this value, the device uses the Use Trusted Relay Point setting from the common device configuration with which this device associates.</td>
</tr>
<tr>
<td>Always Use Prime Line</td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• Off—When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received.</td>
</tr>
<tr>
<td></td>
<td>• On—When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls.</td>
</tr>
<tr>
<td></td>
<td>• Default—Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco CallManager service.</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message</td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• Off—If the phone is idle, pressing the Messages button on the phone automatically dials the voice-messaging system from the line that has a voice message. Unified Communications Manager always selects the first line that has a voice message. If no line has a voice message, the primary line gets used when the phone user presses the Messages button.</td>
</tr>
<tr>
<td></td>
<td>• On—If the phone is idle, the primary line on the phone becomes the active line for retrieving voice messages when the phone user presses the Messages button on the phone</td>
</tr>
<tr>
<td></td>
<td>• Default—Unified Communications Manager uses the configuration from the Always Use Prime Line for Voice Message service parameter, which supports the Cisco CallManager service.</td>
</tr>
</tbody>
</table>
Configure Endpoint Devices

Analog Telephone Adaptor 186 Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>From the drop-down list, choose a geolocation. You can choose the Unspecified geolocation, which designates that this device does not associate with a geolocation. You can also choose a geolocation that has been configured with the System &gt; Geolocation Configuration menu option.</td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td>Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Unified Communications Manager ignores any presentation restriction that is received for internal calls. Use this configuration in combination with the calling line ID presentation and connected line ID presentation configuration at the translation pattern level. Together, these settings allow you to configure call display restrictions to selectively present or block calling and/or connected line display information for each call.</td>
</tr>
<tr>
<td>Allow Control of Device from CTI</td>
<td>Check this check box to allow CTI to control and monitor this device. If the associated directory number specifies a shared line, the check box should be enabled as long as at least one associated device specifies a combination of device type and protocol that CTI supports.</td>
</tr>
<tr>
<td>Logged into Hunt Group</td>
<td>When the CTI port gets added to a hunt list, the administrator can log the user in or out by checking (and unchecking) this check box. Users use the softkey on the phone to log their phone in or out of the hunt list.</td>
</tr>
<tr>
<td>Remote Device</td>
<td>Check this box to allocate a buffer for the device when it registers and to bundle SCCP messages to the phone. <strong>Tip</strong> Because this feature consumes resources, be sure to check this check box only when you are experiencing signaling delays.</td>
</tr>
<tr>
<td>Hot Line Device</td>
<td>Check this check box to make this device a Hotline device. Hotline devices can only connect to other Hotline devices. This feature is an extension of PLAR, which configures a phone to automatically dial one directory number when it goes off-hook. Hotline provides additional restrictions that you can apply to devices that use PLAR. To implement Hotline, you must also create a softkey template without supplementary service softkeys, and apply it to the Hotline device.</td>
</tr>
</tbody>
</table>

Number Presentation Transformation

Table 25: Caller ID for Calls From This Phone

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</td>
</tr>
</tbody>
</table>
### Table 26: Remote Number

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the Trunk Configuration window.</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>From the drop-down list, choose the calling search space (CSS) that contains the calling party transformation pattern that you want to apply on the remote calling number for calls received on this device.</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>Check this check box to apply the Calling Party Transformation CSS configured at the device pool to which this device belongs to transform the remote calling and remote connected number.</td>
</tr>
</tbody>
</table>

### Table 27: Protocol Specific Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| BLF Presence Group            | From the drop-down list, choose a Busy Lamp Field (BLF) presence group for the end user. The selected group specifies the destinations that the end user can monitor  
|                               | The default value for BLF Presence Group specifies Standard Presence group, configured with installation. BLF Presence Groups that are configured in Cisco Unified Administration also appear in the drop-down list. |
| Device Security Profile       | Choose the security profile to apply to the device.  
|                               | You must apply a security profile to all devices that are configured in Unified Communications Manager Administration. |
| SUBSCRIBE Calling Search Space | Supported with the Presence feature, the SUBSCRIBE calling search space determines how Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user.  
|                               | From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list.  
|                               | If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces. |
| Unattended Port               | Check this check box to indicate an unattended port on this device. |
For devices that are running SCCP, check this check box to disable RFC2833 support.

Table 28: Product-Specific Configuration Layout

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model-specific configuration fields that the device manufacturer defines</td>
<td>To view field descriptions and help for product-specific configuration items, click the “?” information icon in the Product Specific Configuration area to display help in a popup dialog box. If you need more information, see the documentation for ATA 186.</td>
</tr>
</tbody>
</table>

Analog Telephone Adaptor 187 Configuration Fields

Table 29: Analog Telephony Adaptor 187 Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MAC Address</td>
<td>Enter the Media Access Control (MAC) address that identifies ATA 187. Make sure that the value comprises 12 hexadecimal characters. You can determine the MAC address for ATA 187 in any of these ways: • Look at the MAC label on the back of ATA 187. • Display the web page for ATA 187 and click the Device Information hyperlink.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a text description of the ATA 187. This field can contain up to 128 characters. You can use all characters except quotes (“”), close angle bracket (&gt;), open angle bracket (&lt;), backslash (), ampersand (&amp;), and percent sign (%).</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Choose the device pool to which you want the ATA 187 assigned. The device pool defines sets of common characteristics for devices, such as region, date/time group, and softkey template. To see the Device Pool configuration settings, click the View Details link.</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>Choose the Common Device Configuration to which you want the ATA 187 assigned. To see the Common Device Configuration settings, click the View Details link.</td>
</tr>
<tr>
<td>Phone Button Template</td>
<td>Choose the appropriate phone button template. The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.</td>
</tr>
<tr>
<td><strong>Field</strong></td>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>---------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Common Phone Profile</td>
<td>From the drop-down list, choose a common phone profile from the list of available common phone profiles. To see the Common Phone Profile settings, click the View Details link.</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>From the drop-down list, choose the calling search space or leave the calling search space as the default of &lt;None&gt;.</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>From the drop-down list, choose the appropriate calling search space for the device to use when it performs automated alternate routing (AAR) or leave the calling search space as the default of &lt;None&gt;.</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Choose the appropriate Media Resource Group List. A Media Resource Group List comprises a prioritized grouping of media resource groups. If you choose &lt;None&gt;, Cisco Unified CM uses the Media Resource Group List that is defined in the device pool.</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>From the drop-down list, choose the audio source to use for music on hold (MOH) when a user initiates a hold action.</td>
</tr>
<tr>
<td>Location</td>
<td>From the drop-down list, choose the location that is associated with the phones and gateways in the device pool.</td>
</tr>
<tr>
<td>AAR Group</td>
<td>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. If no AAR group is specified, Cisco Unified CM uses the AAR group that is associated with Device Pool or Line.</td>
</tr>
<tr>
<td>User Locale</td>
<td>From the drop-down list, choose the user locale that is associated with the CTI Port. The user locale identifies a set of detailed information to support users, including language and font. If you do not specify a user locale, Cisco Unified CM uses the user locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Network Locale</td>
<td>From the drop-down list, choose the network locale that is associated with the CTI Port. The network locale contains a definition of the tones and cadences that the phone in a specific geographic area uses. If you do not specify a network locale, Cisco Unified CM uses the network locale that is associated with the device pool.</td>
</tr>
</tbody>
</table>
| Built in Bridge           | Enable or disable the built-in conference bridge for the barge feature by using the Built In Bridge drop-down list. Choose one of the following: | • On  
• Off  
• Default |
| Privacy                   | For Privacy, choose **On** in the Privacy drop-down list.                                                                                     |
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Mobility Mode</td>
<td>From the drop-down list, turn the device mobility feature on or off for this device or choose Default to use the default device mobility mode. Default setting uses the value for the Device Mobility Mode service parameter for the device.</td>
</tr>
<tr>
<td>Owner</td>
<td>Select User or Anonymous (Public/Shared Space), for the owner type.</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>From the drop-down list, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device. Assigning a user ID to the device also moves the device from “Unassigned Devices” to “Users” in the License Usage Report. <strong>Note</strong> Do not configure this field if you are using extension mobility. Extension mobility does not support device owners.</td>
</tr>
<tr>
<td>Phone Load Name</td>
<td>Enter the custom software for ATA 187.</td>
</tr>
</tbody>
</table>
| Use Trusted Relay Point      | Choose one of the following values:  
  
  - **Off**—Choose this value to disable the use of a Trusted Relay Point (TRP) with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates.  
  
  - **On**—Choose this value to enable the use of a TRP with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates.  
  
  - **Default**—If you choose this value, the device uses the Use Trusted Relay Point setting from the common device configuration with which this device associates. |
| Always Use Prime Line        | From the drop-down list, choose one of the following options:  
  
  - **Off**—When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received.  
  
  - **On**—When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls.  
  
  - **Default**—Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco CallManager service. |
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Always Use Prime Line for Voice Message   | From the drop-down list, choose one of the following options:  
• Off—If the phone is idle, pressing the Messages button on the phone automatically dials the voice-messaging system from the line that has a voice message. Unified Communications Manager always selects the first line that has a voice message. If no line has a voice message, the primary line gets used when the phone user presses the Messages button.  
• On—If the phone is idle, the primary line on the phone becomes the active line for retrieving voice messages when the phone user presses the Messages button on the phone  
• Default—Unified Communications Manager uses the configuration from the Always Use Prime Line for Voice Message service parameter, which supports the Cisco CallManager service.                                                                                                                                                                |
| Geolocation                               | From the drop-down list, choose a geolocation.  
You can choose the Unspecified geolocation, which designates that this device does not associate with a geolocation.  
You can also choose a geolocation that has been configured with the System > Geolocation Configuration menu option.                                                                                                                                                                                                                     |
| Ignore Presentation Indicators (internal calls only) | Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Unified Communications Manager ignores any presentation restriction that is received for internal calls.  
Use this configuration in combination with the calling line ID presentation and connected line ID presentation configuration at the translation pattern level. Together, these settings allow you to configure call display restrictions to selectively present or block calling and/or connected line display information for each call. |
| Logged into Hunt Group                    | When the ATA 187 gets added to a hunt list, the administrator can log the user in or out by checking (and unchecking) this check box.  
Users use the softkey on the phone to log their phone in or out of the hunt list.                                                                                                                                                                                                                                                                                     |
| Remote Device                             | Check this box to allocate a buffer for the device when it registers and to bundle SCCP messages to the phone.  
**Tip** Because this feature consumes resources, be sure to check this check box only when you are experiencing signaling delays.                                                                                                                                                                                                                                                                              |
Check this checkbox to designate a phone as protected, which enables the phone to play a 2-second tone to notify the user when a call is encrypted and both phones are configured as protected devices. The tone plays for both parties when the call is answered. The tone does not play unless both phones are protected and the call occurs over encrypted media.

Checking this check box represents only one of several configuration requirements for the secure indication tone to play. For a detailed description of the secure indication tone feature and the configuration requirements, see the Feature Configuration Guide for Cisco Unified Communications Manager at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html.

If you check this check box and the system determines that the call is not encrypted, the phone plays nonsecure indication tone to alert the user that the call is not protected.

### Number Presentation Transformation

**Table 30: Caller ID for Calls From This Phone**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Protected Device</td>
<td>Check this check box to designate a phone as protected, which enables the</td>
</tr>
<tr>
<td></td>
<td>phone to play a 2-second tone to notify the user when a call is encrypted</td>
</tr>
<tr>
<td></td>
<td>and both phones are configured as protected devices. The tone plays for both</td>
</tr>
<tr>
<td></td>
<td>parties when the call is answered. The tone does not play unless both</td>
</tr>
<tr>
<td></td>
<td>phones are protected and the call occurs over encrypted media.</td>
</tr>
<tr>
<td></td>
<td>Checking this check box represents only one of several configuration</td>
</tr>
<tr>
<td></td>
<td>requirements for the secure indication tone to play. For a detailed</td>
</tr>
<tr>
<td></td>
<td>description of the secure indication tone feature and the configuration</td>
</tr>
<tr>
<td></td>
<td>requirements, see the Feature Configuration Guide for Cisco Unified</td>
</tr>
<tr>
<td></td>
<td>Communications Manager at <a href="http://www.cisco.com/c/en/us/support/unified-">http://www.cisco.com/c/en/us/support/unified-</a></td>
</tr>
<tr>
<td></td>
<td>communications/unified-communications-manager-callmanager/</td>
</tr>
<tr>
<td></td>
<td>products-installation-and-configuration-guides-list.html.</td>
</tr>
<tr>
<td></td>
<td>If you check this check box and the system determines that the call is</td>
</tr>
<tr>
<td></td>
<td>not encrypted, the phone plays nonsecure indication tone to alert the user</td>
</tr>
<tr>
<td></td>
<td>that the call is not protected.</td>
</tr>
</tbody>
</table>

**Table 31: Remote Number**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>This setting allows you to localize the calling party number on the device.</td>
</tr>
<tr>
<td></td>
<td>Make sure that the Calling Party Transformation CSS that you choose</td>
</tr>
<tr>
<td></td>
<td>contains the calling party transformation pattern that you want to</td>
</tr>
<tr>
<td></td>
<td>assign to this device.</td>
</tr>
<tr>
<td>Use Device Pool Calling Party</td>
<td>To use the Calling Party Transformation CSS that is configured in the</td>
</tr>
<tr>
<td>Transformation CSS</td>
<td>device pool that is assigned to this device, check this check box.</td>
</tr>
<tr>
<td></td>
<td>If you do not check this check box, the device uses the Calling Party</td>
</tr>
<tr>
<td></td>
<td>Transformation CSS that you configured in the Trunk Configuration window.</td>
</tr>
<tr>
<td></td>
<td>From the drop-down list, choose the calling search space (CSS) that</td>
</tr>
<tr>
<td></td>
<td>contains the calling party transformation pattern that you want to apply</td>
</tr>
<tr>
<td></td>
<td>on the remote calling number for calls received on this device.</td>
</tr>
<tr>
<td></td>
<td>Check this check box to apply the Calling Party Transformation CSS</td>
</tr>
<tr>
<td></td>
<td>configured at the device pool to which this device belongs to transform</td>
</tr>
<tr>
<td></td>
<td>the remote calling and remote connected number.</td>
</tr>
</tbody>
</table>
### Table 32: Protocol Specific Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>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:</td>
</tr>
<tr>
<td></td>
<td>• <strong>None</strong>—This option, which serves as the default setting, indicates that no packet capturing is occurring. After you complete packet capturing, configure this setting.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Batch Processing Mode</strong>—Cisco Unified CM 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 CM, which stores the file for seven days, also stores the keys that encrypt the file in a secure location. Cisco Unified CM 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.</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>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.</td>
</tr>
<tr>
<td>BLF Presence Group</td>
<td>From the drop-down list, choose a Busy Lamp Field (BLF) presence group for the end user. The selected group specifies the destinations that the end user can monitor. The default value for BLF Presence Group specifies Standard Presence group, configured with installation. BLF Presence Groups that are configured in Cisco Unified Administration also appear in the drop-down list.</td>
</tr>
<tr>
<td>SIP Dial Rules</td>
<td>If required, choose the appropriate SIP dial rule. SIP dial rules provide local dial plans for Cisco Unified IP Phones 7940, and 7960, so users do not have to press a key or wait for a timer before the call gets processed. Leave the SIP Dial Rules field set to &lt;None&gt; if you do not want dial rules to apply to the IP phone that is running SIP. This means that the user must use the Dial softkey or wait for the timer to expire before the call gets processed.</td>
</tr>
<tr>
<td>MTP Preferred Originating Codec</td>
<td>From the drop-down list, choose the codec to use if a media termination point is required for SIP calls.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Choose the security profile to apply to the device. You must apply a security profile to all devices that are configured in Unified Communications Manager Administration.</td>
</tr>
<tr>
<td>Rerouting Calling Search Space</td>
<td>From the drop-down list, choose a calling search space to use for rerouting. The rerouting calling search space of the referrer gets used to find the route to the refer-to target. When the Refer fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message. The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target.</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>Supported with the Presence feature, the SUBSCRIBE calling search space determines how Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user. From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list. If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces.</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Choose the default SIP profile or a specific profile that was previously created. SIP profiles provide specific SIP information for the phone such as registration and keepalive timers, media ports, and do not disturb control.</td>
</tr>
<tr>
<td>Digest User</td>
<td>Choose an end user that you want to associate with the phone for this setting that is used with digest authentication (SIP security). Ensure that you configured digest credentials for the user that you choose, as specified in the End User Configuration window. After you save the phone configuration and apply the configuration update to the phone, the digest credentials for the user get added to the phone configuration file.</td>
</tr>
</tbody>
</table>
### Field: Media Termination Point Required

Use this field to indicate whether a media termination point is used to implement features that ATA 187 does not support (such as hold and transfer).

Check the Media Termination Point Required check box if you want to use an MTP to implement features. Uncheck the Media Termination Point Required check box if you do not want to use an MTP to implement features.

Use this check box only for ATA 187 clients and those ATA 187 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 this device becomes the endpoint of a video call, the call will be audio only.

### Field: Unattended Port

Check this check box to indicate an unattended port on this device.

### Field: Required DTMF Reception

For devices that are running SIP and SCCP, check this check box to require DTMF reception for this phone.

**Note:** In configuring Cisco Unified Mobility features, when using intercluster DNs as remote destinations for an IP phone via SIP trunk (either intercluster trunk [ICT] or gateway), check this check box so that DTMF digits can be received out of band, which is crucial for Enterprise Feature Access midcall features.

---

### Table 33: Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation</td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• No Pending Operation—Displays when no certificate operation is occurring</td>
</tr>
<tr>
<td></td>
<td>(default setting).</td>
</tr>
<tr>
<td></td>
<td>• Install/Upgrade—Installs a new or upgrades an existing locally significant</td>
</tr>
<tr>
<td></td>
<td>certificate in the phone.</td>
</tr>
<tr>
<td></td>
<td>• Delete—Deletes the locally significant certificate that exists in the phone.</td>
</tr>
<tr>
<td></td>
<td>• Troubleshoot—Retrieves the locally significant certificate (LSC) or the</td>
</tr>
<tr>
<td></td>
<td>manufacture installed certificate (MIC), so you can view the certificate</td>
</tr>
<tr>
<td></td>
<td>credentials in the CAPF trace file. If both certificate types exist in</td>
</tr>
<tr>
<td></td>
<td>the phone, Cisco Unified CM creates two files, one for each certificate</td>
</tr>
<tr>
<td></td>
<td>type. By choosing the Troubleshooting option, you can verify that an</td>
</tr>
<tr>
<td></td>
<td>LSC or MIC exists in the phone.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Authentication Mode</td>
<td>This field allows you to choose the authentication method that the phone uses during the CAPF certificate operation. From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• By Authentication String—Installs/upgrades, deletes, or troubleshoots a locally significant certificate only when the user enters the CAPF authentication string on the phone.</td>
</tr>
<tr>
<td></td>
<td>• By Null String—Installs/upgrades, deletes, or troubleshoots a locally significant certificate without user intervention. This option provides no security; Cisco strongly recommends that you choose this option only for closed, secure environments.</td>
</tr>
<tr>
<td></td>
<td>• By Existing Certificate (Precedence to LSC)—Installs/upgrades, deletes, or troubleshoots a locally significant certificate if a manufacture-installed certificate (MIC) or locally significant certificate (LSC) exists in the phone. If a LSC exists in the phone, authentication occurs via the LSC, regardless whether a MIC exists in the phone. If a MIC and LSC exist in the phone, authentication occurs via the LSC. If a LSC does not exist in the phone, but a MIC does exist, authentication occurs via the MIC.</td>
</tr>
<tr>
<td></td>
<td>Before you choose this option, verify that a certificate exists in the phone. If you choose this option and no certificate exists in the phone, the operation fails.</td>
</tr>
<tr>
<td></td>
<td>At any time, the phone uses only one certificate to authenticate to CAPF even though a MIC and LSC can exist in the phone at the same time. If the primary certificate, which takes precedence, becomes compromised for any reason, or, if you want to authenticate via the other certificate, you must update the authentication mode.</td>
</tr>
<tr>
<td></td>
<td>• By Existing Certificate (Precedence to MIC)—Installs, upgrades, deletes, or troubleshoots a locally significant certificate if a LSC or MIC exists in the phone. If a MIC exists in the phone, authentication occurs via the MIC, regardless whether a LSC exists in the phone. If a LSC exists in the phone, but a MIC does not exist, authentication occurs via the LSC.</td>
</tr>
<tr>
<td></td>
<td>Before you choose this option, verify that a certificate exists in the phone. If you choose this option and no certificate exists in the phone, the operation fails.</td>
</tr>
<tr>
<td>Note</td>
<td>The CAPF settings that are configured in the Phone Security Profile window interact with the CAPF parameters that are configured in the Phone Configuration window.</td>
</tr>
<tr>
<td>Authentication String</td>
<td>If you chose the By Authentication String option in the Authentication Mode drop-down list, this field applies. Manually enter a string or generate a string by clicking the Generate String button. Ensure that the string contains 4 to 10 digits.</td>
</tr>
<tr>
<td></td>
<td>To install, upgrade, delete, or troubleshoot a locally significant certificate, the phone user or administrator must enter the authentication string on the phone.</td>
</tr>
</tbody>
</table>
### Field Description

**Key Size (Bits)**  
For this setting that is used for CAPF, choose the key size for the certificate from the drop-down list. The default setting equals 1024. Other options include 512 and 2048.

If you choose a higher key size than the default setting, the phones take longer to generate the entropy that is required to generate the keys. Key generation, which is set at low priority, allows the phone to function while the action occurs. Depending on the phone model, you may notice that key generation takes up to 30 or more minutes to complete.

**Note**  
The CAPF settings that are configured in the Phone Security Profile window interact with the CAPF parameters that are configured in the Phone Configuration window.

**Operation Completes by**  
This field, which supports the Install/Upgrade, Delete, and Troubleshoot Certificate Operation options, specifies the date and time in which you must complete the operation.

The values that display apply for the publisher database server.

**Certificate Operation Status**  
This field displays the progress of the certificate operation; for example, `<operation type>` pending, failed, or successful, where operating type equals the Install/Upgrade, Delete, or Troubleshoot Certificate Operation options. You cannot change the information that displays in this field.

### Table 34: Secure Shell User

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User</td>
<td>Enter a user ID for the secure shell user. You can enter any alphanumeric or special characters up to 50 characters. Invalid characters include &quot;, %, &amp;, &lt;, &gt;, and . This field displays when the phone device that you are configuring supports SSH access. Cisco Technical Assistance Center (TAC) uses secure shell for troubleshooting and debugging. Contact TAC for further assistance. See the <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html">Security Guide for Cisco Unified Communications Manager</a> at this release for information about how to configure encrypted phone configuration files to ensure that Cisco Unified CM does not send SSH credentials to the phone in the clear.</td>
</tr>
<tr>
<td>Secure Shell Password</td>
<td>Enter the password for a secure shell user. You can enter any alphanumeric or special characters up to 200 characters. Invalid characters include &quot;, %, &amp;, &lt;, &gt;, and . Contact TAC for further assistance. See the <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html">Security Guide for Cisco Unified Communications Manager</a> at this release for information about how to configure encrypted phone configuration files to ensure that Cisco Unified CM does not send SSH credentials to the phone in the clear.</td>
</tr>
</tbody>
</table>
Table 35: Product-Specific Configuration Layout

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model-specific configuration fields that the device manufacturer defines</td>
<td>To view field descriptions and help for product-specific configuration items, click the “?” information icon in the Product Specific Configuration area to display help in a popup dialog box. If you need more information, see the documentation for ATA 187.</td>
</tr>
</tbody>
</table>

Analog Telephony Adaptor 190 Configuration Fields

Table 36: Analog Telephony Adaptor 190 Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| MAC Address               | Enter the Media Access Control (MAC) address that identifies ATA 190. Make sure that the value comprises 12 hexadecimal characters. You can determine the MAC address for ATA 190 in any of these ways:  
  • Look at the MAC label on the back of ATA 190.  
  • Display the web page for ATA 190 and click the Device Information hyperlink. |
| Description               | Enter a text description of the ATA 190. This field can contain up to 128 characters. You can use all characters except quotes ("), close angle bracket (>), open angle bracket (<), backslash (\), ampersand (&), and percent sign (%). |
| Device Pool               | Choose the device pool to which you want the ATA 190 assigned. The device pool defines sets of common characteristics for devices, such as region, date/time group, and softkey template.  
  To see the Device Pool configuration settings, click the View Details link. |
| Common Device Configuration | Choose the Common Device Configuration to which you want the ATA 190 assigned.  
  To see the Common Device Configuration settings, click the View Details link. |
| Phone Button Template     | Choose the appropriate phone button template. The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button. |
| Common Phone Profile      | From the drop-down list, choose a common phone profile from the list of available common phone profiles.  
  To see the Common Phone Profile settings, click the View Details link. |
<p>| Calling Search Space      | From the drop-down list, choose the calling search space or leave the calling search space as the default of &lt;None&gt;. |</p>
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR Calling Search Space</td>
<td>From the drop-down list, choose the appropriate calling search space for the device to use when it performs automated alternate routing (AAR) or leave the calling search space as the default of <code>&lt;None&gt;</code>.</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Choose the appropriate Media Resource Group List. A Media Resource Group List comprises a prioritized grouping of media resource groups. If you choose <code>&lt;None&gt;</code>, Cisco Unified CM uses the Media Resource Group List that is defined in the device pool.</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>From the drop-down list, choose the audio source to use for music on hold (MOH) when a user initiates a hold action.</td>
</tr>
<tr>
<td>Location</td>
<td>From the drop-down list, choose the location that is associated with the phones and gateways in the device pool.</td>
</tr>
<tr>
<td>AAR Group</td>
<td>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. If no AAR group is specified, Cisco Unified CM uses the AAR group that is associated with Device Pool or Line.</td>
</tr>
<tr>
<td>User Locale</td>
<td>From the drop-down list, choose the user locale that is associated with the CTI Port. The user locale identifies a set of detailed information to support users, including language and font. If you do not specify a user locale, Cisco Unified CM uses the user locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Network Locale</td>
<td>From the drop-down list, choose the network locale that is associated with the CTI Port. The network locale contains a definition of the tones and cadences that the phone in a specific geographic area uses. If you do not specify a network locale, Cisco Unified CM uses the network locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Built in Bridge</td>
<td>Enable or disable the built-in conference bridge for the barge feature by using the Built In Bridge drop-down list. Choose one of the following:</td>
</tr>
<tr>
<td></td>
<td>- On</td>
</tr>
<tr>
<td></td>
<td>- Off</td>
</tr>
<tr>
<td></td>
<td>- Default</td>
</tr>
<tr>
<td>Privacy</td>
<td>For Privacy, choose On in the Privacy drop-down list.</td>
</tr>
<tr>
<td>Device Mobility Mode</td>
<td>From the drop-down list, turn the device mobility feature on or off for this device or choose Default to use the default device mobility mode. Default setting uses the value for the Device Mobility Mode service parameter for the device.</td>
</tr>
<tr>
<td>Owner</td>
<td>Select User or Anonymous (Public/Shared Space), for the owner type.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------------</td>
<td>------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Owner User ID                | From the drop-down list, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device. Assigning a user ID to the device also moves the device from “Unassigned Devices” to “Users” in the License Usage Report.  
  **Note** Do not configure this field if you are using extension mobility. Extension mobility does not support device owners. |
<p>| Phone Load Name              | Enter the custom software for ATA 190.                                                                                                                                                                     |
| Use Trusted Relay Point      | Choose one of the following values:                                                                                                                                                                         |
|                              | • <strong>Off</strong>—Choose this value to disable the use of a Trusted Relay Point (TRP) with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates. |
|                              | • <strong>On</strong>—Choose this value to enable the use of a TRP with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates. |
|                              | • <strong>Default</strong>—If you choose this value, the device uses the Use Trusted Relay Point setting from the common device configuration with which this device associates.                                                |
| Always Use Prime Line        | From the drop-down list, choose one of the following options:                                                                                                                                              |
|                              | • <strong>Off</strong>—When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received.                                                              |
|                              | • <strong>On</strong>—When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls. |
|                              | • <strong>Default</strong>—Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco CallManager service.                                             |
| Always Use Prime Line for Voice Message | From the drop-down list, choose one of the following options:                                                                                                                                              |
|                              | • <strong>Off</strong>—If the phone is idle, pressing the Messages button on the phone automatically dials the voice-messaging system from the line that has a voice message. Unified Communications Manager always selects the first line that has a voice message. If no line has a voice message, the primary line gets used when the phone user presses the Messages button. |
|                              | • <strong>On</strong>—If the phone is idle, the primary line on the phone becomes the active line for retrieving voice messages when the phone user presses the Messages button on the phone.                                                       |
|                              | • <strong>Default</strong>—Unified Communications Manager uses the configuration from the Always Use Prime Line for Voice Message service parameter, which supports the Cisco CallManager service.                                             |</p>
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>From the drop-down list, choose a geolocation. You can choose the Unspecified geolocation, which designates that this device does not associate with a geolocation. You can also choose a geolocation that has been configured with the <strong>System &gt; Geolocation Configuration</strong> menu option.</td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td>Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Unified Communications Manager ignores any presentation restriction that is received for internal calls. Use this configuration in combination with the calling line ID presentation and connected line ID presentation configuration at the translation pattern level. Together, these settings allow you to configure call display restrictions to selectively present or block calling and/or connected line display information for each call.</td>
</tr>
<tr>
<td>Logged into Hunt Group</td>
<td>When the ATA 190 gets added to a hunt list, the administrator can log the user in or out by checking (and unchecking) this check box. Users use the softkey on the phone to log their phone in or out of the hunt list.</td>
</tr>
<tr>
<td>Remote Device</td>
<td>Check this box to allocate a buffer for the device when it registers and to bundle SCCP messages to the phone. <strong>Tip</strong> Because this feature consumes resources, be sure to check this check box only when you are experiencing signaling delays.</td>
</tr>
<tr>
<td>Protected Device</td>
<td>Check this check box to designate a phone as protected, which enables the phone to play a 2-second tone to notify the user when a call is encrypted and both phones are configured as protected devices. The tone plays for both parties when the call is answered. The tone does not play unless both phones are protected and the call occurs over encrypted media. Checking this check box represents only one of several configuration requirements for the secure indication tone to play. For a detailed description of the secure indication tone feature and the configuration requirements, see the <em>Security Guide for Cisco Unified Communications Manager</em> at <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html">http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html</a>. If you check this check box and the system determines that the call is not encrypted, the phone plays nonsecure indication tone to alert the user that the call is not protected.</td>
</tr>
</tbody>
</table>
### Number Presentation Transformation

**Table 37: Caller ID for Calls From This Phone**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the Trunk Configuration window.</td>
</tr>
</tbody>
</table>

**Table 38: Remote Number**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>From the drop-down list, choose the calling search space (CSS) that contains the calling party transformation pattern that you want to apply on the remote calling number for calls received on this device.</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>Check this check box to apply the Calling Party Transformation CSS configured at the device pool to which this device belongs to transform the remote calling and remote connected number.</td>
</tr>
</tbody>
</table>

**Table 39: Protocol Specific Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>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:</td>
</tr>
<tr>
<td></td>
<td>• <strong>None</strong>—This option, which serves as the default setting, indicates that no packet capturing is occurring. After you complete packet capturing, configure this setting.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Batch Processing Mode</strong>—Cisco Unified CM 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 CM, which stores the file for seven days, also stores the keys that encrypt the file in a secure location. Cisco Unified CM 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.</td>
</tr>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| 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. |
| BLF Presence Group                         | From the drop-down list, choose a Busy Lamp Field (BLF) presence group for the end user. The selected group specifies the destinations that the end user can monitor.  
The default value for BLF Presence Group specifies Standard Presence group, configured with installation. BLF Presence Groups that are configured in Cisco Unified Administration also appear in the drop-down list. |
| SIP Dial Rules                             | If required, choose the appropriate SIP dial rule. SIP dial rules provide local dial plans for Cisco Unified IP Phones 7940, and 7960, so users do not have to press a key or wait for a timer before the call gets processed.  
Leave the SIP Dial Rules field set to <None> if you do not want dial rules to apply to the IP phone that is running SIP. This means that the user must use the Dial softkey or wait for the timer to expire before the call gets processed. |
| MTP Preferred Originating Codec            | From the drop-down list, choose the codec to use if a media termination point is required for SIP calls.                                                                                                     |
| Device Security Profile                    | Choose the security profile to apply to the device.  
You must apply a security profile to all devices that are configured in Unified Communications Manager Administration.                                                                                           |
| Rerouting Calling Search Space             | From the drop-down list, choose a calling search space to use for rerouting.  
The rerouting calling search space of the referrer gets used to find the route to the refer-to target. When the Refer fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message.  
The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target. |
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| SUBSCRIBE Calling Search Space| Supported with the Presence feature, the SUBSCRIBE calling search space determines how Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user.  

From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list.  

If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces. |
| SIP Profile                   | Choose the default SIP profile or a specific profile that was previously created. SIP profiles provide specific SIP information for the phone such as registration and keepalive timers, media ports, and do not disturb control. |
| Digest User                   | Choose an end user that you want to associate with the phone for this setting that is used with digest authentication (SIP security).  

Ensure that you configured digest credentials for the user that you choose, as specified in the End User Configuration window.  

After you save the phone configuration and apply the configuration update to the phone, the digest credentials for the user get added to the phone configuration file.  

| Media Termination Point Required| Use this field to indicate whether a media termination point is used to implement features that ATA 190 does not support (such as hold and transfer).  

Check the Media Termination Point Required check box if you want to use an MTP to implement features. Uncheck the Media Termination Point Required check box if you do not want to use an MTP to implement features.  

Use this check box only for ATA 190 clients and those ATA 190 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 this device becomes the endpoint of a video call, the call will be audio only. |
| Unattended Port               | Check this check box to indicate an unattended port on this device. |
For devices that are running SIP and SCCP, check this check box to require DTMF reception for this phone.

**Note** In configuring Cisco Unified Mobility features, when using intercluster DNs as remote destinations for an IP phone via SIP trunk (either intercluster trunk [ICT] or gateway), check this check box so that DTMF digits can be received out of band, which is crucial for Enterprise Feature Access midcall features.

### Table 40: Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation</td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• No Pending Operation—Displays when no certificate operation is occurring</td>
</tr>
<tr>
<td></td>
<td>(default setting).</td>
</tr>
<tr>
<td></td>
<td>• Install/Upgrade—Installs a new or upgrades an existing locally significant</td>
</tr>
<tr>
<td></td>
<td>certificate in the phone.</td>
</tr>
<tr>
<td></td>
<td>• Delete—Deletes the locally significant certificate that exists in the phone.</td>
</tr>
<tr>
<td></td>
<td>• Troubleshoot—Retrieves the locally significant certificate (LSC) or the</td>
</tr>
<tr>
<td></td>
<td>manufacture installed certificate (MIC), so you can view the certificate</td>
</tr>
<tr>
<td></td>
<td>credentials in the CAPF trace file. If both certificate types exist in the</td>
</tr>
<tr>
<td></td>
<td>phone, Cisco Unified CM creates two trace files, one for each certificate</td>
</tr>
<tr>
<td></td>
<td>type.</td>
</tr>
<tr>
<td></td>
<td>By choosing the Troubleshooting option, you can verify that an LSC or</td>
</tr>
<tr>
<td></td>
<td>MIC exists in the phone.</td>
</tr>
</tbody>
</table>

This field allows you to choose the authentication method that the phone uses during the CAPF certificate operation.

From the drop-down list, choose one of the following options:

- **By Authentication String**—Installs/upgrades, deletes, or troubleshoots a locally significant certificate only when the user enters the CAPF authentication string on the phone.

- **By Null String**—Installs/upgrades, deletes, or troubleshoots a locally significant certificate without user intervention.

  This option provides no security; Cisco strongly recommends that you choose this option only for closed, secure environments.

- **By Existing Certificate (Precedence to LSC)**—Installs/upgrades, deletes, or troubleshoots a locally significant certificate if a manufacture-installed certificate (MIC) or locally significant certificate (LSC) exists in the phone. If a LSC exists in the phone, authentication occurs via the LSC, regardless whether a MIC exists in the phone. If a MIC and LSC exist in the phone, authentication occurs via the LSC. If a LSC does not exist in the phone, but a MIC does exist, authentication occurs via the MIC.

  Before you choose this option, verify that a certificate exists in the phone. If you choose this option and no certificate exists in the phone, the operation fails.

  At any time, the phone uses only one certificate to authenticate to CAPF even though a MIC and LSC can exist in the phone at the same time. If the primary certificate, which takes precedence, becomes compromised for any reason, or, if you want to authenticate via the other certificate, you must update the authentication mode.

- **By Existing Certificate (Precedence to MIC)**—Installs, upgrades, deletes, or troubleshoots a locally significant certificate if a LSC or MIC exists in the phone. If a MIC exists in the phone, authentication occurs via the MIC, regardless whether a LSC exists in the phone. If a LSC exists in the phone, but a MIC does not exist, authentication occurs via the LSC.

  Before you choose this option, verify that a certificate exists in the phone. If you choose this option and no certificate exists in the phone, the operation fails.

**Note** The CAPF settings that are configured in the Phone Security Profile window interact with the CAPF parameters that are configured in the Phone Configuration window.

**Authentication String**

If you chose the **By Authentication String** option in the **Authentication Mode** drop-down list, this field applies. Manually enter a string or generate a string by clicking the **Generate String** button. Ensure that the string contains 4 to 10 digits.

To install, upgrade, delete, or troubleshoot a locally significant certificate, the phone user or administrator must enter the authentication string on the phone.
For this setting that is used for CAPF, choose the key size for the certificate from the drop-down list. The default setting equals 1024. Other options include 512 and 2048.

If you choose a higher key size than the default setting, the phones take longer to generate the entropy that is required to generate the keys. Key generation, which is set at low priority, allows the phone to function while the action occurs. Depending on the phone model, you may notice that key generation takes up to 30 or more minutes to complete.

**Note**  The CAPF settings that are configured in the Phone Security Profile window interact with the CAPF parameters that are configured in the Phone Configuration window.

### Table 41: Secure Shell User

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User</td>
<td>Enter a user ID for the secure shell user. You can enter any alphanumeric or special characters up to 50 characters. Invalid characters include &quot;,&quot;, &quot;,&quot;, &quot;,&quot;, &quot;,&quot;, and &quot;.&quot;. This field displays when the phone device that you are configuring supports SSH access. Cisco Technical Assistance Center (TAC) uses secure shell for troubleshooting and debugging. Contact TAC for further assistance. See the Security Guide for Cisco Unified Communications Manager at <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html">http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html</a> for this release for information about how to configure encrypted phone configuration files to ensure that Cisco Unified CM does not send SSH credentials to the phone in the clear.</td>
</tr>
</tbody>
</table>
Secure Shell Password

Enter the password for a secure shell user. You can enter any alphanumeric or special characters up to 200 characters. Invalid characters include "", %, &, <, >, and \. Contact TAC for further assistance.


Table 42: Product-Specific Configuration Layout

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model-specific configuration fields that the device manufacturer defines</td>
<td>To view field descriptions and help for product-specific configuration items, click the “?” information icon in the Product Specific Configuration area to display help in a popup dialog box. If you need more information, see the documentation for ATA 190.</td>
</tr>
</tbody>
</table>
Configure Software-Based Endpoints

- Software-Based Endpoint Configuration, on page 311
- Configure CTI Ports, on page 311
- Configure an H.323 Client, on page 319
- Configure Cisco IP Communicator, on page 320

Software-Based Endpoint Configuration

Complete the tasks in this chapter to configure software-based endpoints such as CTI ports, H.323 clients, and Cisco IP Communicator.

Configure CTI Ports

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.

Step 2  Click Add New.

Step 3  From the Phone Type drop down list, select CTI Port and click Next. The Phone Configuration window appears.

Step 4  Configure the fields in the Phone Configuration window. See the Related Topics section for more information about the fields and their configuration options.

Step 5  Click Save.

Related Topics
- CTI Port Settings, on page 312
## CTI Port Settings

**Table 43: CTI Port Settings**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Specifies the name for the CTI Port that is automatically populated based on the Owner User ID. The format of the device name is <code>CTIRD&lt;OwnerUserID&gt;</code> by default. This field is editable. The device name can comprise up to 15 characters. Valid characters include letters, numbers, dashes, dots (periods), spaces, and underscores.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a text description of the CTI Port. This field can contain up to 128 characters. You can use all characters except quotes (&quot;), close angle bracket (&gt;), open angle bracket (&lt;), backslash (), ampersand (&amp;), and percent sign (%).</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Choose the device pool to which you want the CTI Port assigned. The device pool defines sets of common characteristics for devices, such as region, date/time group, and softkey template. To see the Device Pool configuration settings, click the View Details link.</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>Choose the Common Device Configuration to which you want the CTI Port assigned. To see the Common Device Configuration settings, click the View Details link.</td>
</tr>
<tr>
<td>Common Phone Profile</td>
<td>From the drop-down list box, choose a common phone profile from the list of available common phone profiles. To see the Common Phone Profile settings, click the View Details link.</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>From the drop-down list, choose the calling search space or leave the calling search space as the default of &lt;None&gt;.</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>From the drop-down list, choose the appropriate calling search space for the device to use when it performs automated alternate routing (AAR) or leave the calling search space as the default of &lt;None&gt;.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Choose the appropriate Media Resource Group List. A Media Resource Group List comprises a prioritized grouping of media resource groups. If you choose <code>&lt;None&gt;</code>, Cisco Unified CM uses the Media Resource Group List that is defined in the device pool.</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>From the drop-down list, choose the audio source to use for music on hold (MOH) when a user initiates a hold action.</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>From the drop-down list, choose the audio source to use for MOH when the network initiates a hold action.</td>
</tr>
<tr>
<td>Location</td>
<td>From the drop-down list, choose the location that is associated with the phones and gateways in the device pool.</td>
</tr>
<tr>
<td>AAR Group</td>
<td>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. If no AAR group is specified, Cisco Unified CM uses the AAR group that is associated with Device Pool or Line.</td>
</tr>
<tr>
<td>User Locale</td>
<td>From the drop-down list box, choose the user locale that is associated with the CTI Port. The user locale identifies a set of detailed information to support users, including language and font. If no user locale is specified, Cisco Unified CM uses the user locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Network Locale</td>
<td>From the drop-down list box, choose the network locale that is associated with the CTI Port. The network locale contains a definition of the tones and cadences that the phone in a specific geographic area uses. If no network locale is specified, Cisco Unified CM uses the user locale that is associated with the device pool.</td>
</tr>
<tr>
<td>Privacy</td>
<td>For Privacy, choose On in the Privacy drop-down list box.</td>
</tr>
<tr>
<td>Owner</td>
<td>Select User or Anonymous (Public/Shared Space, for the owner type.</td>
</tr>
</tbody>
</table>
### Owner User ID

From the drop-down list box, choose the user ID of the assigned CTI Port user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device. Assigning a user ID to the device also moves the device from “Unassigned Devices” to “Users” in the License Usage Report.

### Join Across Lines

From the drop-down list box, enable or disable the Join Across Lines feature for this device or choose Default to use the service parameter setting.

### Use Trusted Relay Point

Choose one of the following values:

- **Off**—Choose this value to disable the use of a Trusted Relay Point (TRP) with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates.

- **On**—Choose this value to enable the use of a TRP with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates.

- **Default**—If you choose this value, the device uses the Use Trusted Relay Point setting from the common device configuration with which this device associates.

### Always Use Prime Line

From the drop-down list box, choose one of the following options:

- **Off**—When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received.

- **On**—When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls.

- **Default**—Cisco Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco CallManager service.
From the drop-down list box, choose one of the following options:

- **Off**—If the phone is idle, pressing the Messages button on the phone automatically dials the voice-messaging system from the line that has a voice message. Cisco Unified Communications Manager always selects the first line that has a voice message. If no line has a voice message, the primary line gets used when the phone user presses the Messages button.

- **On**—If the phone is idle, the primary line on the phone becomes the active line for retrieving voice messages when the phone user presses the Messages button on the phone.

- **Default**—Cisco Unified Communications Manager uses the configuration from the Always Use Prime Line for Voice Message service parameter, which supports the Cisco CallManager service.

From the drop-down list box, choose a geolocation.

You can choose the Unspecified geolocation, which designates that this device does not associate with a geolocation.

You can also choose a geolocation that has been configured with the **System > Geolocation Configuration** menu option.

Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Cisco Unified Communications Manager ignores any presentation restriction that is received for internal calls.

Use this configuration in combination with the calling line ID presentation and connected line ID presentation configuration at the translation pattern level. Together, these settings allow you to configure call display restrictions to selectively present or block calling and/or connected line display information for each call.

When the CTI port gets added to a hunt list, the administrator can log the user in or out by checking (and unchecking) this check box.

Users use the softkey on the phone to log their phone in or out of the hunt list.
### Number Presentation Transformation

**Table 44: Caller ID for Calls From This Phone**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Remote Device</strong></td>
<td>Check this box to allocate a buffer for the device when it registers and to bundle SCCP messages to the phone.</td>
</tr>
<tr>
<td><strong>Tip</strong></td>
<td>Because this feature consumes resources, be sure to check this check box only when you are experiencing signaling delays.</td>
</tr>
</tbody>
</table>

**Tip**

Remote Device Number Presentation Transformation

*Table 44: Caller ID for Calls From This Phone*

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Calling Party Transformation CSS</strong></td>
<td>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</td>
</tr>
<tr>
<td><strong>Use Device Pool Calling Party Transformation CSS</strong></td>
<td>To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the Trunk Configuration window.</td>
</tr>
</tbody>
</table>

**Table 45: Remote Number**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Calling Party Transformation CSS</strong></td>
<td>From the drop-down list box, choose the calling search space (CSS) that contains the calling party transformation pattern that you want to apply on the remote calling number for calls received on this device.</td>
</tr>
<tr>
<td><strong>Use Device Pool Calling Party Transformation CSS</strong></td>
<td>Check this check box to apply the Calling Party Transformation CSS configured at the device pool to which this device belongs to transform the remote calling and remote connected number.</td>
</tr>
</tbody>
</table>
### Table 46: Protocol Specific Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLF Presence Group</td>
<td>From the drop-down list box, choose a Busy Lamp Field (BLF) presence group for the end user. The selected group specifies the destinations that the end user can monitor. The default value for BLF Presence Group specifies Standard Presence group, configured with installation. BLF Presence Groups that are configured in Cisco Unified Administration also appear in the drop-down list box.</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Choose the security profile to apply to the device. You must apply a security profile to all devices that are configured in Cisco Unified Communications Manager Administration.</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user. From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list. If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces.</td>
</tr>
<tr>
<td>Unattended Port</td>
<td>Check this check box to indicate an unattended port on this device.</td>
</tr>
</tbody>
</table>
### Table 47: MLPP and Confidential Access Level Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
<td>From the drop-down list, choose an Multilevel Precedence and Preemption (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.</td>
</tr>
</tbody>
</table>
| Confidential Access Mode | From the drop-down list box, select one of the following options to set the Confidential Access Level mode:  
  • Fixed—Confidential Access Level value has higher precedence over call completion.  
  • Variable—Call completion has higher precedence over CAL level. |
| Confidential Access Level | Select the appropriate Confidential Access Level value from the drop-down list box. |

### Table 48: Do Not Disturb Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
<td>Check this check box to enable Do Not Disturb on the remote device.</td>
</tr>
<tr>
<td>DND Option</td>
<td>When you enable DND, Call Reject option specifies that no incoming call information gets presented to the user. Depending on how you configure the DND Incoming Call Alert parameter, the device may play a beep or display a flash notification of the call.</td>
</tr>
</tbody>
</table>
When you enable the DND Ringer Off or Call Reject option, this parameter specifies how a call displays on the device.

From the drop-down list, choose one of the following options:

- **None**—This option specifies that the DND Incoming Call Alert setting from the Common Phone Profile window gets used for this device.
- **Disable**—This option disables both beep and flash notification of a call, but, for the DND Ringer Off option, incoming call information still gets displayed. For the DND Call Reject option, no call alerts display, and no information gets sent to the device.
- **Beep Only**—For an incoming call, this option causes the device to play a beep tone only.
- **Flash Only**—For an incoming call, this option causes the device to display a flash alert.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| DND Incoming Call Alert | When you enable the DND Ringer Off or Call Reject option, this parameter specifies how a call displays on the device. From the drop-down list, choose one of the following options:
- None—This option specifies that the DND Incoming Call Alert setting from the Common Phone Profile window gets used for this device.
- Disable—This option disables both beep and flash notification of a call, but, for the DND Ringer Off option, incoming call information still gets displayed. For the DND Call Reject option, no call alerts display, and no information gets sent to the device.
- Beep Only—For an incoming call, this option causes the device to play a beep tone only.
- Flash Only—For an incoming call, this option causes the device to display a flash alert. |

## Configure an H.323 Client

### Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2** Click **Add New**.

**Step 3** From the **Phone Type** drop down list, select H.323 Client and click **Next**. The **Phone Configuration** window appears.

**Step 4** Configure the fields in the **Phone Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

**Step 5** Click **Save**.

### Related Topics

[H.323 Client Settings](#), on page 320
H.323 Client Settings

Configure Cisco IP Communicator

Cisco IP Communicator is a software-based application that allows users to place and receive phone calls by using their personal computers. It provides the same functionality as a full-featured Cisco Unified IP Phone. Cisco IP Communicator depends upon the Cisco Unified Communications Manager call-processing system to provide telephony features and voice-over-IP capabilities. You administer Cisco IP Communicator as a phone device by using the Cisco Unified Communications Manager Administration Phone Configuration window.

Procedure

Step 1
From Cisco Unified CM Administration, choose Device > Phone.
The Find and List Phones window appears.

Step 2
Click Add New.

Step 3
From the Phone Type drop down list, select Cisco IP Communicator and click Next.

Step 4
From the Select the device protocol drop-down list, select either SCCP or SIP and click Next.
The Phone Configuration window appears.

Step 5
Configure the following mandatory fields in the Phone Configuration window.

- **Device Name**—enter a name to identify the Cisco IP Communicator device.
- **Device Pool**—choose the device pool to which you want this phone assigned. The device pool defines sets of common characteristics for devices, such as region, date/time group, and softkey template.
- **Phone Button Template**—choose the appropriate phone button template. The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.
- **Owner User ID**—from the drop-down list box, choose the user ID of the assigned phone user.
- **Device Security Profile**—choose the security profile to apply to the device.

You can use default configuration for the remaining fields. See the online help for more information about the fields and their configuration options.

Step 6
Click Save.

Step 7
In the Association area, click Line [1] - Add a new DN.

Step 8
In the Directory Number field, enter the directory number that you want to associate with the phone.

Step 9
Click Save.
CHAPTER 42

Configure Cisco IP Phones

- Cisco IP Phones Overview, on page 321
- Cisco IP Phones Configuration Task Flow, on page 321

Cisco IP Phones Overview

Cisco IP Phones are full-featured telephones that provide voice communication over an IP network. To provide this capability, the IP phones interact with several other key Cisco Unified IP Telephony and network components, such as Unified Communications Manager, DNS and DHCP servers, TFTP servers, media resources, Cisco Power over Ethernet (PoE), and others. These IP phones function much like digital business phones that allow you to place and receive phone calls and to access features such as mute, hold, transfer, speed dial, call forward, and more. In addition, because Cisco IP Phones connect to your data network, they offer enhanced IP telephony features, such as access to network information and services and customizable features and services. The phones also support security features that include file authentication, device authentication, signaling encryption, and media encryption.

This chapter describes how to configure a phone to make it operational on your system. To configure features such as Call Park, Call Forward, Busy Lamp Field (BLF), Call Pickup, and Speed Dial, see the Feature Configuration Guide for Cisco Unified Communications Manager at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html

Cisco IP Phones Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Phones, on page 322</td>
<td>Perform this task to configure a SIP or an SCCP Phone.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure EnergyWise, on page 327</td>
<td>To reduce power consumption, configure the phone to power down (sleep) and power up (wake) automatically.</td>
</tr>
</tbody>
</table>
Configure Phones

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 3</strong> Configure a Client Services Framework Device, on page 329</td>
<td>Perform this procedure to configure a Client Services Framework device. A Client Services Framework device can be any of the following:</td>
</tr>
<tr>
<td></td>
<td>- Cisco Unified Communications Integration for Microsoft Office Communicator</td>
</tr>
<tr>
<td></td>
<td>- Cisco Unified Communications Integration for WebEx Connect</td>
</tr>
<tr>
<td></td>
<td>- Cisco Unified Personal Communicator (Release 8.0 and later)</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure a CTI Remote Device, on page 331</td>
<td>Perform this procedure to configure a CTI Remote Device. A CTI remote device is a device type that represents off-cluster phones that users can use with Cisco UC applications. The device type is configured with one or more lines (directory numbers) and one or more remote destinations.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Migrate Phone Data, on page 341</td>
<td>Perform this procedure if you are migrating from one phone to another and you do not need to use the old phone anymore.</td>
</tr>
</tbody>
</table>

Configure Phones

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> To configure SIP phones, perform the following procedures:</td>
<td>Perform these procedures if you have phones that use Session Initiation Protocol (SIP). SIP provides the primary interface between the phone and other network components. In addition to SIP, other protocols are used for various functions such as DHCP for IP address assignment, DNS for domain name to address resolution, and TFTP for downloading image and configuration data. For detailed steps about configuring a VPN client, see the Feature Configuration Guide for Cisco Unified Communications Manager at <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html">http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html</a>.</td>
</tr>
<tr>
<td>- Configure SIP Phone Secure Port, on page 323</td>
<td></td>
</tr>
<tr>
<td>- Restart Services, on page 323</td>
<td></td>
</tr>
<tr>
<td>- Configure SIP Profile, on page 324</td>
<td></td>
</tr>
<tr>
<td>- Configure Phone Security Profile, on page 324</td>
<td></td>
</tr>
<tr>
<td>- Configure a Phone, on page 325</td>
<td></td>
</tr>
<tr>
<td>- Configure Cisco IP Phone Services, on page 326</td>
<td></td>
</tr>
<tr>
<td>- Configure a VPN client.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> To configure SCCP Phones, perform the following procedures:</td>
<td>Perform these procedures if you want to configure Cisco IP Phones that use the Skinny Client Control Protocol (SCCP). SCCP uses</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>
What to do next

Provide power, verify network connectivity, and configure network settings for the Cisco Unified IP Phone. For more information about configuring network settings, see the *Cisco Unified IP Phone Administration Guide* for your Cisco Unified IP Phone model.

**Configure SIP Phone Secure Port**

Follow these steps to configure the SIP Phone Secure Port. Cisco Unified Communications Manager uses this port to listen to SIP phones for SIP line registrations over TLS.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose System &gt; Cisco Unified CM.</td>
</tr>
<tr>
<td>Step 2</td>
<td>In the Cisco Unified Communications Manager TCP Port Settings for this Server section, specify a port number in the SIP Phone Secure Port field, or leave the field set to default. The default value is 5061.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Apply Config.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Ok.</td>
</tr>
</tbody>
</table>

**Restart Services**

Follow these steps to restart Cisco CallManager and Cisco CTL Provider services.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From the Cisco Unified Serviceability interface, choose Tools &gt; Control Center - Feature Services.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Choose the Cisco Unified Communications Manager server from the Servers drop-down list. In the CM Services area, Cisco CallManager displays in the Service Name column.</td>
</tr>
</tbody>
</table>
Configure SIP Profile

Use this procedure to configure SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks.

Before you begin

• Configure SIP Phone Secure Port, on page 323
• Restart Services, on page 323

Procedure

Step 1  In Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.
Step 2  Click Find.
Step 3  For the profile that you want to copy, click the file icon in the Copy column.
Step 4  Enter the name and description of the new profile.
Step 5  If you have the IPv6 stack configured and you are deploying two stacks, check the Enable ANAT check box.
       Note This configuration applies whether you have Unity Connection deployed or not.
Step 6  Click Save.

What to do next

Configure Phone Security Profile, on page 324

Configure Phone Security Profile

Cisco Unified Communications Manager provides a set of predefined, nonsecure profiles for autoregistration. If you want to enable security features for a phone, you must configure a new security profile and apply it to the phone. Follow these steps to configure a new security profile:

Before you begin

If you are configuring SIP phones, complete the following procedures:

• Configure SIP Phone Secure Port, on page 323
• Restart Services, on page 323
• Configure SIP Profile, on page 324
Procedure

Step 1  From Cisco Unified CM Administration, choose System > Security > Phone Security Profile.
Step 2  Click Add New.
Step 3  From the Phone Security Profile Type drop-down list, choose the Universal Device Template to create a profile that you can use when provisioning through the device templates.

Note  Optionally, you can also create security profiles for specific device models.

Step 4  Select the protocol.
Step 5  Enter an appropriate name for the profile in the Name field.
Step 6  If you want to use TLS signaling to connect to the device, set the Device Security Mode to Authenticated or Encrypted and the Transport Type to TLS.
Step 7  (Optional) Check the Enable OAuth Authentication check box if you want the phone to use digest authentication.
Step 8  (Optional) Check the TFTP Encrypted Config check box if you want to use encrypted TFTP.
Step 9  Complete the remaining fields in the Phone Security Profile Configuration window. For help with the fields and their settings, see the online help.
Step 10  Click Save.

Configure a Phone

Perform these steps to manually add the phone to the Cisco Unified Communications Manager database. You do not have to perform these steps if you are using autoregistration. If you opt for autoregistration, Cisco Unified Communications Manager automatically adds the phone and assigns the directory number. For more information about enabling autoregistration, see Configure Autoregistration Task Flow, on page 517.

Before you begin

- Configure Phone NTP References, on page 49
- Configure Phone Security Profile, on page 324
- Add a Date/Time Group, on page 50
- Set Up SIP Dial Rule, on page 167 (when configuring a SIP phone)

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Click Add New.
Step 3  From the Phone Type drop-down list, select the appropriate Cisco IP Phone model.
Step 4  Click Next.
Step 5  From the Select the device protocol drop-down list, choose one of the following:
  - SCCP
  - SIP
Step 6  Click Next.
Step 7  Configure the fields in the **Phone Configuration** window. See the online help for more information about the fields and their configuration options.

**Note**  The CAPF settings that are configured in the security profile relate to the Certificate Authority Proxy Function settings that display in the Phone Configuration window. You must configure CAPF settings for certificate operations that involve manufacturer-installed certificates (MICs) or locally significant certificates (LSC). See the Cisco Unified Communications Manager Security Guide for more information about how CAPF settings that you update in the phone configuration window affect security profile CAPF settings.

Step 8  Click **Save**.
Step 9  In the **Association** area, click **Line [1] - Add a new DN**.
Step 10  In the **Directory Number** field, enter the directory number that you want to associate with the phone.
Step 11  Click **Save**.

---

**What to do next**

For SIP or SCCP phones:

Configure Cisco IP Phone Services, on page 326

---

**Configure Cisco IP Phone Services**

Configure services for Cisco IP Phones if you want to provide phone services such as a company directory, visual voicemail, or weather forecasts to the Cisco IP Phones. Cisco provides certain default IP phone services, which install automatically with Cisco Unified Communications Manager. You can also create customized Cisco IP Phone services for your site. Follow these steps to configure the customized services in Unified Communications Manager.

**Before you begin**

Configure a Phone, on page 325

**Procedure**

Step 1  From Cisco Unified CM Administration, choose **Device > Device Settings > Phone Services**.
Step 2  Click **Add New**.
Step 3  Configure the fields in the **IP Phone Services Configuration** window. See the online help for more information about the fields and their configuration options.
Step 4  Click **Save**.

**What to do next**

- Add services to the phones in the database if they are not classified as enterprise subscriptions. You can add services to the phones using Bulk Administration Tool (BAT) or Cisco Unified Communications Self Care Portal. For more information, see *Cisco Unified Communications Manager Bulk Administration System Configuration Guide for Cisco Unified Communications Manager, Release 11.5(1)SU1*.

• You can assign the services to the phone buttons, if the phone model supports these buttons. For more information about assigning services, see Cisco IP Phone User Guide for your phone model.

• Configure VPN Client (optional).

Configure EnergyWise

Before you begin

• Ensure that your system includes an EnergyWise controller. For example, a Cisco Switch with the EnergyWise feature enabled.

• See the user documentation for your phone model to check whether your phone model supports the EnergyWise feature.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.

Step 2 Specify the search criteria and click Find.

A list of phones that are configured on the Cisco Unified Communications Manager is displayed.

Step 3 Choose the phone for which you want to configure the EnergyWise feature.

Step 4 Configure the EnergyWise fields in the Product Specific Configuration Layout section. See the Related Topics section for more information about the fields and their configuration options.

Step 5 Click Save.

Related Topics

EnergyWise Configuration Fields, on page 327

EnergyWise Configuration Fields

Table 49: EnergyWise Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Power Save Plus</td>
<td>Select the day for which you want the phone to automatically power off. You can select multiple days by pressing and holding the Control key while clicking on the days for the schedule. By default, no days are selected.</td>
</tr>
</tbody>
</table>
### EnergyWise Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phone On Time</td>
<td>Enter the time in 24 hour format, where 00:00 is midnight. This value determines when the phone automatically powers on for the days selected in the Enable Power Save Plus field.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> To wake up the phone before the <strong>Phone On Time</strong>, you must power on the phone from the switch. For more information, see the switch documentation.</td>
</tr>
<tr>
<td>Phone Off Time</td>
<td>Enter the time in 24 hour format, where 00:00 is midnight. This value determines when the phone automatically powers down for the days selected in the Enable Power Save Plus field. If the <strong>Phone On Time</strong> and the <strong>Phone Off Time</strong> fields contain the same value, the phone does not power down.</td>
</tr>
<tr>
<td>Phone Off Idle Timeout</td>
<td>Specify the duration for which the phone must be idle before the phone powers down. You can specify any value from 20 minutes to 1440 minutes. The default value is 60 minutes.</td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td>Check this check box to instruct the phone to play an audible alert 10 minutes, 7 minutes, 4 minutes, and 30 seconds before the time specified in the <strong>Phone Off Time</strong> field. This check box applies only if the Enable Power Save Plus list box has one or more days selected.</td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td>Specify the EnergyWise domain that the phone is in. The maximum length allowed is 127 characters.</td>
</tr>
<tr>
<td>EnergyWise Secret</td>
<td>Specify the security secret password that is used to communicate with the endpoints in the EnergyWise domain. The maximum length allowed is 127 characters.</td>
</tr>
<tr>
<td>Allow EnergyWise Overrides</td>
<td>Check this check box to disable Power Save Plus. If you check this check box, the EnergyWise domain controller policy overrides the <strong>Power On Time</strong> and <strong>Power Off Time</strong> values.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> Leaving the Allow EnergyWise Overrides check box checked with no days selected in the Enable Power Save Plus field does not disable Power Save Plus.</td>
</tr>
</tbody>
</table>
Configure a Client Services Framework Device

Perform this procedure to configure a Client Services Framework device. A Client Services Framework device can be any of the following:

- Cisco Unified Communications Integration for Microsoft Office Communicator
- Cisco Unified Communications Integration for WebEx Connect
- Cisco Unified Personal Communicator (Release 8.0 and later)

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Add a Client Services Framework Device, on page 329</td>
<td>Add a device that uses the Client Services Framework.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate Device with End User, on page 330</td>
<td>Associate an end user account to the Client Services Framework device.</td>
</tr>
</tbody>
</table>

Add a Client Services Framework Device

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Add New**.
3. From the **Phone Type** drop-down list, choose **Cisco Unified Client Services Framework**.
4. Click **Next**.
5. Configure the fields in the **Phone Configuration** window. See the Related Topics section for more information about the fields and their configuration options.
6. Click **Save**.
7. In the **Association** area, click **Line [1] - Add a new DN**.
8. In the **Directory Number** field, enter the directory number that you want to associate with the client services framework device.
9. Click **Save**.

**Related Topics**

- Client Services Framework Device Configuration Fields, on page 330
### Client Services Framework Device Configuration Fields

#### Table 50: Client Services Framework Device Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Enter a name to identify the Client Services Framework device. The name can include up to 15 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_). <strong>Note</strong> When you configure the device name for the Cisco Unified Personal Communicator, make sure that the name starts with UPC.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a brief description for the device. The description can include up to 50 characters in any language, but it cannot include double-quotes (&quot;), percentage sign (%), ampersand (&amp;), back-slash (), or angle brackets (&lt;&gt;).</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Choose the device pool to which you want this device assigned.</td>
</tr>
<tr>
<td>Phone Button Template</td>
<td>Choose <strong>Standard Client Services Framework</strong>.</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>Choose the user ID of the assigned Client Services Framework device user. The user ID is recorded in the call detail record (CDR) for all calls made from this device.</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Choose <strong>Cisco Unified Client Services Framework - Standard SIP Non-secure Profile</strong>.</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Choose <strong>Standard SIP Profile</strong>.</td>
</tr>
</tbody>
</table>

### Associate Device with End User

Use this procedure to associate an end user with the Client Services Framework Device.

**Procedure**

1. From Cisco Unified CM Administration, choose **User Management > End User**.
2. Click **Find** and select the user whom you want to associate to the device.
3. In the **Device Information** section, click **Device Association**. The User Device Association window appears.
4. Click **Find** to view a list of available devices.
5. Select the device that you want to associate, and click **Save Selected/Changes**.
6. From **Related Links**, choose **Back to User**, and click **Go**.
The End User Configuration window appears, and the associated device that you chose appears in the Controlled Devices pane.

## Configure CTI Remote Device

### Purpose

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure a CTI Remote Device, on page 331</td>
<td>Create a CTI Remote Device.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Add a Directory Number to a Device, on page 335</td>
<td>To register a CTI Remote Device, you must add a Directory Number to that device.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure a Remote Destination, on page 335</td>
<td>Configure the remote destinations that you want to associate with the CTI remote device.</td>
</tr>
</tbody>
</table>

### Configure a CTI Remote Device

#### Procedure

1. From Cisco Unified CM Administration, choose Device > Phone.
2. Click Add New.
3. From the Phone Type drop down list, select CTI Remote Device and click Next.
4. Configure the fields in the Phone Configuration window. See the Related Topics section for more information about the fields and their configuration options.
5. Click Save.

#### Related Topics

- CTI Remote Device Configuration Fields, on page 331

### CTI Remote Device Configuration Fields

#### CTI Remote Device Information

**Table 51: Device Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Specifies the registration status of the CTI Remote Device.</td>
</tr>
<tr>
<td>Device Status</td>
<td>Specifies if the device is active or inactive.</td>
</tr>
<tr>
<td>Device Trust</td>
<td>Specifies if the device is trusted.</td>
</tr>
</tbody>
</table>
## CTI Remote Device Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Active Remote Destination</td>
<td>Specifies if the remote destination which is active. The CTI client can specify one remote destination as 'active' at any one given time. Incoming calls and Dial via Office (DVO) calls are routed to the active remote destination.</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>From the drop-down list, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device.</td>
</tr>
<tr>
<td>Device Name</td>
<td>Specifies the name for the CTI Remote Device that is automatically populated based on the Owner User ID.</td>
</tr>
<tr>
<td></td>
<td>The format of the device name is <code>CTIRD&lt;OwnerUserID&gt;</code> by default.</td>
</tr>
<tr>
<td></td>
<td>This field is editable. The device name can comprise up to 15 characters. Valid characters include letters, numbers, dashes, dots (periods), spaces, and underscores.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a text description of the CTI remote device.</td>
</tr>
<tr>
<td></td>
<td>This field can contain up to 128 characters. You can use all characters except quotes (&quot;), close angle bracket (&gt;), open angle bracket (&lt;), backslash (), ampersand (&amp;), and percent sign (%).</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Select the device pool which defines the common characteristics for CTI remote devices.</td>
</tr>
<tr>
<td></td>
<td>For more information on how to configure the device pool, see Device Pool Configuration Settings.</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>From the drop-down list, choose the calling search space or leave the calling search space as the default of &quot;&lt;None&gt;&quot;.</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>From the drop-down list, choose the audio source to use for music on hold (MOH) when a user initiates a hold action.</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>From the drop-down list, choose the audio source to use for MOH when the network initiates a hold action.</td>
</tr>
<tr>
<td>Location</td>
<td>From the drop-down list, choose the location that is associated with the phones and gateways in the device pool.</td>
</tr>
</tbody>
</table>
This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device pool.

Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Cisco Unified CM ignores any presentation restriction that is received for internal calls.

### Call Routing Information

**Table 52: Inbound/Outbound Calls Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the Trunk Configuration window.</td>
</tr>
</tbody>
</table>

**Table 53: Protocol Specific Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presence Group</td>
<td>Configure this field with the Presence feature. If you are not using this application user with presence, leave the default (None) setting for presence group. From the drop-down list, choose a Presence group for the application user. The group selected specifies the destinations that the application user, such as IPMAsysUser, can monitor.</td>
</tr>
</tbody>
</table>
Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user.

From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list.

If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces.

From the drop-down list, choose a calling search space to use for rerouting.

The rerouting calling search space of the referrer is used to find the route to the refer-to target. When the Refer message fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message.

The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user. From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list. If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces.</td>
</tr>
<tr>
<td>Rerouting Calling Search Space</td>
<td>From the drop-down list, choose a calling search space to use for rerouting. The rerouting calling search space of the referrer is used to find the route to the refer-to target. When the Refer message fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message. The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
<td>Check this check box to enable Do Not Disturb on the remote device.</td>
</tr>
<tr>
<td>DND Option</td>
<td>When you enable DND on the phone, Call Reject option specifies that no incoming call information gets presented to the user. Depending on how you configure the DND Incoming Call Alert parameter, the phone may play a beep or display a flash notification of the call.</td>
</tr>
</tbody>
</table>

Table 54: Do Not Disturb Information
Add a Directory Number to a Device

To register a CTI Remote Device, you must add a Directory Number to that device. You cannot register a CTI Remote Device without a Directory Number. You can add a maximum of five Directory Numbers to the CTI Remote Device.

Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone**.

**Step 2** Specify the filter criteria and click on the CTI Remote Device for which you want to associate the directory number.

**Step 3** In the **Association** pane, click **Add a new DN link**.

**Step 4** Configure the fields in the **Directory Number Configuration** window. See the online help for more information about the fields and their configuration options.

**Step 5** Click **Save**.

Configure a Remote Destination

You can configure one or more remote destinations for a CTI Remote Device. A remote destination is a mobile or other phone that you can configure to perform remote destination pickup (accept transfers from the desk phone of the user) and accept incoming Cisco Unified Mobility calls. The remote destination that is associated with the CTI Remote Device specifies the phone number to reach the remote device. The maximum number of remote destinations that you can configure for a CTI Remote Device is dependent on the Remote Destination limit set for the Owner User ID.

Remote destinations can include any of the following devices:

- Single-mode mobile (cellular) phones
- Smartphones
- Dual-mode phones
- Enterprise IP phones that are not in the same cluster as the desk phone
- Home phone numbers in the PSTN

Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone > CTI Remote Device > Associated Remote Destinations**.

**Step 2** Specify the filter criteria and click on the CTI Remote Device for which you want to configure the remote destination.

**Step 3** In the **Associated Remote Destinations** pane, select **Add a New Remote Destination**. Alternatively, you can also use the **Device > Phone > Add New** menu path to configure a remote destination.

**Step 4** Configure the fields in the **Remote Destination Configuration** window. See the Related Topics section for more information about the fields and their configuration options.
Step 5  
Click Save.

Related Topics
Remote Destination Configuration Fields, on page 336

Remote Destination Configuration Fields

Table 55: Remote Destination Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>Enter the name of the remote destination.</td>
</tr>
<tr>
<td>Destination Number</td>
<td>Enter the number that you would dial from within the enterprise. Include the area code and any additional digits that are required to reach an outside line. The maximum field length is 24 characters; individual characters can take the values 0-9, *, #, and +. Cisco recommends that you configure the caller ID of the remote destination.</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>From the drop-down list, choose the owner of the remote destination.</td>
</tr>
<tr>
<td>Enable Unified Mobility features</td>
<td>Check the check box to enable Unified Mobility features.</td>
</tr>
<tr>
<td>Remote Destination Profile</td>
<td>From drop-down list, choose the profile that you configured.</td>
</tr>
<tr>
<td>Enable Single Number Reach</td>
<td>Check the check box to enable Single-Number_Reach for the remote destination.</td>
</tr>
<tr>
<td>Enable Move to Mobile</td>
<td>This is an optional field. If your phone is a mobile phone, check this check box.</td>
</tr>
</tbody>
</table>

Configure a Cisco Spark Remote Device

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure a Cisco Spark Remote Device, on page 337</td>
<td>Create a Cisco Spark remote device.</td>
</tr>
<tr>
<td>Step 2 Add a Directory Number to a Cisco Spark Device, on page 341</td>
<td>To register a Cisco Spark remote device, you must add a Directory Number to that device.</td>
</tr>
</tbody>
</table>
Configure a Cisco Spark Remote Device

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Click Add New.
Step 3  From the Phone Type drop down list, select Cisco Spark Remote Device and click Next.
Step 4  Configure the fields in the Phone Configuration window. See the Related Topics section for more information about the fields and their configuration options.
Step 5  Click Save.

Related Topics
Cisco Spark Remote Device Configuration Fields, on page 337

Cisco Spark Remote Device Configuration Fields

Table 56: Cisco Webex Remote Device Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Information</td>
<td></td>
</tr>
<tr>
<td>Registration</td>
<td>Specifies the registration status of the Cisco Webex remote device.</td>
</tr>
<tr>
<td>Device Status</td>
<td>Specifies if the device is active or inactive.</td>
</tr>
<tr>
<td>Device Trust</td>
<td>Specifies whether the device is trusted or not trusted.</td>
</tr>
<tr>
<td>Active Remote Destination</td>
<td>Specifies if the remote destination is active. The Cisco Webex client has only one active remote destination by default. All incoming calls are routed to the active remote destination. This field is to none even though it is associated to an active remote destination.</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>From the drop-down list, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device.</td>
</tr>
</tbody>
</table>
### Cisco Spark Remote Device Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Device Name             | Specifies the name for the Cisco Webex remote device that is automatically populated based on the Owner User ID.  
The format of the device name is SparkRD<OwnerUserID> by default. The default device name SparkRD must not be changed.  
This field is editable. The device name can comprise up to 15 characters. Valid characters include letters, numbers, dashes, dots (periods), spaces, and underscores. |
| Description             | Enter a text description of the Cisco Webex remote device.  
This field can contain up to 128 characters. You can use all characters except quotes ("), close angle bracket (>), open angle bracket (<), backslash (\), ampersand (&), and percent sign (%). |
| Device Pool             | Select the device pool which defines the common characteristics for Cisco Webex remote devices.  
For more information on how to configure the device pool, see Device Pool Configuration Settings. |
| Calling Search Space    | From the drop-down list, choose the calling search space or leave the calling search space as the default of <None>. |
| User Hold MOH Audio Source | From the drop-down list, choose the audio source to use for music on hold (MOH) when you initiate a hold action.  
**Caution** MOH is currently not supported for Cisco Spark remote device as Hold/Resume feature on the device is not implemented. |
| Network Hold MOH Audio Source | From the drop-down list, choose the audio source to use for MOH when the network initiates a hold action.  
**Caution** MOH is currently not supported for Cisco Spark remote device as Hold/Resume feature on the device is not implemented. |
<p>| Location                | From the drop-down list, choose the location that is associated with the phones and gateways in the device pool. |</p>
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td>Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Cisco Unified CM ignores any presentation restriction that is received for internal calls.</td>
</tr>
</tbody>
</table>

### Call Routing Information

### Inbound and Outbound Calls Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the <strong>Trunk Configuration</strong> window.</td>
</tr>
</tbody>
</table>

### Protocol Specific Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presence Group</td>
<td>Configure this field with the Presence feature. If you are not using this application user with presence, leave the default (None) setting for presence group. From the drop-down list, choose a Presence group for the application user. The group selected specifies the destinations that the application user, such as IPMASysUser, can monitor. <strong>Caution</strong> Presence Group is currently not supported for Cisco Spark remote device.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| SUBSCRIBE Calling Search Space | Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user.  
From the drop-down list, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list.  
If you do not select a different calling search space for the end user 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 can configure a calling search space as you do all calling search spaces.  
**Caution**    
SUBSCRIBE Calling Search Space is currently not supported for Cisco Spark remote device. |
| Rerouting Calling Search Space | From the drop-down list, choose a calling search space to use for rerouting.  
The rerouting calling search space of the referrer is used to find the route to the refer-to target. When the Refer message fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message.  
The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target. |
| Do Not Disturb Information    |                                                                                                                                                                                                                                                                              |
| Do Not Disturb                | Check this check box to enable Do Not Disturb on the remote device.  
**Caution**  
The calls are not routed to Cisco Spark clients if the DND option is enabled.  
**Caution**  
Do Not Disturb is currently not supported for Cisco Spark remote device. |
When you enable DND on the phone, Call Reject option specifies that no incoming call information gets presented to the user. Depending on how you configure the DND Incoming Call Alert parameter, the phone may play a beep or display a flash notification of the call.

**Caution**  Do Not Disturb is currently not supported for Cisco Spark remote device.

### Add a Directory Number to a Cisco Spark Device

To register a Cisco Webex remote device, add a Directory Number to that device. You cannot register a Cisco Webex remote device without a Directory Number. You can add a maximum of five Directory Numbers to the Cisco Webex remote device.

**Procedure**

1. **Step 1**  From Cisco Unified CM Administration, choose Device > Phone.
2. **Step 2**  Specify the filter criteria and click the Cisco Spark Remote Device for which you want to associate the directory number.
3. **Step 3**  In the Association pane, click Add a new DN link.
4. **Step 4**  Configure the fields in the **Directory Number Configuration** window. See the online help for more information about the fields and their configuration options.
5. **Step 5**  Click Save.

### Migrate Phone Data

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Create a Phone Template, on page 342</td>
<td>Create a phone template in Bulk Administration Tool (BAT) for the phone model and protocol to which you want to migrate the data.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Migrate Phone Data, on page 342</td>
<td>Migrate phone data to a different phone.</td>
</tr>
</tbody>
</table>
Create a Phone Template

Procedure

Step 1  From Cisco Unified CM Administration, choose Bulk Administration > Phones > Phone Template.
Step 2  Click Add New.
The Add a New Phone Template window displays.
Step 3  From the Phone Type drop-down list, choose the phone model for which you are creating the template. Click Next.
Step 4  From the Select the Device Protocol drop-down list, choose the device protocol. Click Next.
The Phone Template Configuration window appears with fields and default entries for the chosen device type.
Step 5  Configure the fields in the Phone Template Configuration window. See the online help for more information about the fields and their configuration options.
Step 6  Click Save.

Migrate Phone Data

Before you begin

- Unplug the phone from the network.
- Ensure that there are enough device license units for the new phone.
- Ensure that the phone model supports phone migration.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Specify the search criteria and click Find.
Step 3  Choose and click the phone configuration that you want to migrate.
Step 4  Choose Migrate Phone from the Related Links drop-down list.
The Phone Migration Configuration window appears.
Step 5  From the drop-down list, choose the phone template for the phone model to which you want to migrate the phone configuration.
Step 6  Enter the Media Access Control (MAC) address for the new Cisco Unified IP Phone to which you are migrating the configuration. The MAC address must contain 12 hexadecimal characters.
Step 7  (Optional) Enter a description for the new phone. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).
Step 8  Click Save.
Step 9  If a warning displays that the new phone may lose feature functionality, click OK.
What to do next

Plug the new phone into the network and register the device.
CHAPTER 43

Configure Diagnostics and Reporting for Cisco Unified IP Phones

- Diagnostics and Reporting Overview, on page 345
- Prerequisites, on page 346
- Diagnostics and Reporting Configuration Task Flow, on page 347

Diagnostics and Reporting Overview

Cisco Unified Communications Manager offers two options for ensuring call quality on Cisco IP Phones:

- Call Diagnostics—Call diagnostics includes generating Call Management Records (CMR) and voice quality metrics.
- Quality Report Tool QRT—QRT is a voice-quality and general problem-reporting tool for Cisco Unified IP Phones. This tool allows users to easily and accurately report audio and other general problems with their IP phone.

Call Diagnostics Overview

You can configure Cisco IP Phones that are running SCCP and SIP to collect call diagnostics. Call diagnostics comprises Call Management Records (CMR), also called diagnostic records, and voice quality metrics.

Voice quality metrics are enabled by default and supported on most of the Cisco IP Phones. Cisco IP Phones calculate voice quality metrics based on MOS (Mean Opinion Square) value. Voice quality metrics do not account for noise or distortion, only frame loss.

The CMR records store information about the quality of the streamed audio of the call. You can configure the Unified Communications Manager to generate CMRs. This information is useful for post-processing activities such as generating billing records and network analysis.

Quality Report Tool Overview

The Quality Report Tool (QRT) is a voice-quality and general problem-reporting tool for Cisco IP Phones. This tool allows users to easily and accurately report audio and other general problems with their IP phone.
As a system administrator, you can enable QRT functionality by configuring and assigning a softkey template to display the QRT softkey on a user IP phone. You can choose from two different user modes, depending on the level of user interaction that you want with QRT. You then define how the feature works in your system by configuring system parameters and setting up Cisco Unified Serviceability tools. You can create, customize, and view phone problem reports by using the QRT Viewer application.

When users experience problems with their IP phones, they can report the type of problem and other relevant statistics by pressing the QRT softkey on the Cisco IP Phones during the On Hook or Connected call states. Users can then choose the reason code that best describes the problem that is being reported for the IP phone. A customized phone problem report provides you with the specific information.

QRT attempts to collect the streaming statistics after a user selects the type of problem by pressing the QRT softkey. A call should be active for a minimum of 5 seconds for QRT to collect the streaming statistics.

**Prerequisites**

**Call Diagnostics Prerequisites**

Check if your Cisco Unified IP Phone supports Call Diagnostics.

Use this table to determine if your phone supports Call Diagnostics. The Support for Call Diagnostics legend is as follows:

- X—Supported by phones that are running both SCCP and SIP
- S—SCCP feature only

<table>
<thead>
<tr>
<th>Device</th>
<th>Support for Call Diagnostics</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 7906</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7911</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7921</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7931</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7940</td>
<td>S</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7941</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7942-G</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7942-G/GE</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7945</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7960</td>
<td>S</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7961</td>
<td>X</td>
</tr>
</tbody>
</table>
### Quality Report Tool Prerequisites

Any Cisco IP Phone that includes the following capabilities:

- Support for softkey templates
- Support for IP phone services
- Controllable by CTI
- Contains an internal HTTP server

For more information, see the guide for your phone model.

### Diagnostics and Reporting Configuration Task Flow

#### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Call Diagnostics, on page 348</td>
<td>Perform this task to configure Cisco Unified Communications Manager to generate CMRs. The CMR records store information about the quality of the streamed audio of the call. For more information about accessing CMRs, see the Cisco Unified Communications Manager Call Detail Records Administration Guide. Voice Quality Metrics are automatically enabled on the Cisco IP Phones. For more information about accessing voice quality metrics, see the Cisco Unified IP Phone Administration Guide for your phone model.</td>
</tr>
<tr>
<td>Step 2</td>
<td>To Configure the Quality Report Tool, on page 348, perform the following subtasks: Configure the Quality Report Tool (QRT) so that users who experience problems with their</td>
<td></td>
</tr>
</tbody>
</table>
Configure Call Diagnostics

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Service Parameters.

Step 2  From the Server drop-down list, choose the server on which the Cisco CallManager service is running.

Step 3  From the Service drop-down list, choose Cisco CallManager. The Service Parameter Configuration window appears.

Step 4  In the Clusterwide Parameters (Device - General) area, configure the Call Diagnostics Enabled service parameter. The following options are available:

- **Disabled**—CMRs are not generated.

- **Enabled Only When CDR Enabled Flag is True**—CMRs are generated only when the Call Detail Records (CDR) Enabled Flag service parameter is set to True.

- **Enabled Regardless of CDR Enabled Flag**—CMRs are generated regardless of the CDR Enabled Flag service parameter value.

*Note*  Generating CMRs without enabling the CDR Enabled Flag service parameter can cause uncontrolled disk space consumption. Cisco recommends that you enable CDRs when CMRs are enabled.

Step 5  Click Save.

Configure the Quality Report Tool

Configure the Quality Report Tool (QRT) so that users who experience problems with their IP phones can report the type of problem and other relevant statistics by pressing a QRT softkey.
Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>You must configure the On Hook and Connected call states for the QRT Softkey. The following call states are also available:</td>
</tr>
</tbody>
</table>
| Configure a Softkey Template with the QRT Softkey, on page 349 | • Connected Conference  
• Connected Transfer |
| **Step 2** | To make the softkey template available to phones, you must complete either this step or the following step. Follow this step if your system uses a Common Device Configuration to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones. |
| (Optional) To Associate a QRT Softkey Template with a Common Device Configuration, on page 350, perform the following subtasks: | |
| • Add a QRT Softkey Template to a Common Device Configuration, on page 351  
• Associate a Common Device Configuration with a Phone, on page 352 | |
| **Step 3** | Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in conjunction with the Common Device Configuration. Use this procedure in conjunction with the Common Device Configuration if you need assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment. |
| (Optional) Add the QRT Softkey Template to a Phone, on page 352 | |
| **Step 4** | To Configure QRT in Cisco Unified Serviceability, on page 352, perform the following subtasks: |
| To Configure QRT in Cisco Unified Serviceability, on page 352 | • Activate the Cisco Extended Functions Service, on page 353  
• Configure Alarms, on page 353  
• Configure Traces, on page 354 |
| **Step 5** | (Optional) Configure the Service Parameters for the Quality Report Tool, on page 355 |

**Configure a Softkey Template with the QRT Softkey**

You must configure the On Hook and Connected call states for the QRT Softkey. The following call states are also available:

• Connected Conference

• Connected Transfer
**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Device > Device Settings > Softkey Template**.

**Step 2**
Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

**Step 3**
Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

**Step 4**
Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

**Note** If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

**Step 5**
Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

**Step 6**
From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

**Step 7**
From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

**Step 8**
Repeat the previous step to display the softkey in additional call states.

**Step 9**
Click **Save**.

**Step 10**
Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see **Add a Softkey Template to a Common Device Configuration** and **Associate a Softkey Template with a Phone** sections.

**What to do next**
Perform one of the following steps:

- Add a QRT Sofkey Template to a Common Device Configuration, on page 351
- Add the QRT Sofkey Template to a Phone, on page 352

**Associate a QRT Softkey Template with a Common Device Configuration**

Optional. There are two ways to associate a sofkey template with a phone:

- Add the softkey template to the Phone Configuration.
• Add the softkey template to the Common Device Configuration.

The procedures in this section describe how to associate the softkey template with a Common Device Configuration. Follow these procedures if your system uses a Common Device Configuration to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.

To use the alternative method, see Add the QRT Softkey Template to a Phone, on page 352.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add a QRT Softkey Template to a Common Device Configuration, on page 351</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Common Device Configuration with a Phone, on page 352</td>
</tr>
</tbody>
</table>

Add a QRT Softkey Template to a Common Device Configuration

**Before you begin**
Configure a Softkey Template with the QRT Softkey, on page 349

Procedure

**Step 1**  From Cisco Unified CM Administration, choose Device > Device Settings > Common Device Configuration.

**Step 2**  Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click Add New.
b) Enter a name for the Common Device Configuration in the Name field.
c) Click Save.

**Step 3**  Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click Find and enter the search criteria.
b) Click an existing Common Device Configuration.

**Step 4**  In the Softkey Template drop-down list, choose the softkey template that contains the softkey that you want to make available.

**Step 5**  Click Save.

**Step 6**  Perform one of the following tasks:

• If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.
• If you created a new Common Device Configuration, associate the configuration with devices and then restart them.
**What to do next**

Associate a Common Device Configuration with a Phone, on page 352

**Associate a Common Device Configuration with a Phone**

**Before you begin**

Add a QRT Softkey Template to a Common Device Configuration, on page 351

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose Device > Phone.
- **Step 2** Click Find and select the phone device to add the softkey template.
- **Step 3** From the Common Device Configuration drop-down list, choose the common device configuration that contains the new softkey template.
- **Step 4** Click Save.
- **Step 5** Click Reset to update the phone settings.

**Add the QRT Softkey Template to a Phone**

**Before you begin**

Configure a Softkey Template with the QRT Softkey, on page 349

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose Device > Phone.
- **Step 2** Click Find to display the list of configured phones.
- **Step 3** Choose the phone to which you want to add the phone button template.
- **Step 4** In the Phone Button Template drop-down list, choose the phone button template that contains the new feature button.
- **Step 5** Click Save.
  A dialog box is displayed with a message to press Reset to update the phone settings.

**Configure QRT in Cisco Unified Serviceability**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Activate the Cisco Extended Functions Service, on page 353</td>
<td>Activate the Cisco Extended Functions Service to provide support for voice-quality features such as the Quality Report Tool.</td>
</tr>
</tbody>
</table>
Configure Endpoint Devices

### Activate the Cisco Extended Functions Service

Activate the Cisco Extended Functions Service to provide support for voice-quality features such as the Quality Report Tool.

**Procedure**

- **Step 1**  
  From Cisco Unified Serviceability, choose **Tools > Service Activation**.

- **Step 2**  
  From the **Server** drop-down list, choose the node on which you want to activate the Cisco Extended Functions service.

- **Step 3**  
  Check the **Cisco Extended Functions** check box.

- **Step 4**  
  Click **Save**.

**What to do next**

Configure Alarms, on page 353

### Configure Alarms

Configure alarms for the QRT to log errors in the Application Logs within SysLog Viewer. This function logs alarms, provides a description of the alarms, and recommended actions. You can access the SysLog Viewer from the Cisco Unified Real-Time Monitoring Tool.

**Before you begin**

Activate the Cisco Extended Functions Service, on page 353

**Procedure**

- **Step 1**  
  From the Cisco Unified Serviceability, choose **Alarm > Configuration**.
Configure Traces

Configure traces for the QRT to log trace information for your voice application. After configure the information that you want to include in the trace files for the QRT, you can collect and view trace files by using the Trace and Log Central option in the Cisco Unified Real-Time Monitoring Tool.

Before you begin

Configure Alarms, on page 353

Procedure

Step 1 From Cisco Unified Serviceability, choose Trace > Configuration.
Step 2 From the Server drop-down list, choose the node on which you want to configure traces.
Step 3 From the Service Group drop-down list, choose CM Services.
Step 4 From the Service drop-down list, choose Cisco Extended Functions.
Step 5 Check the Trace On check box.
Step 6 From the Debug Trace Level drop-down list, choose one of the following options:
  • Error—Traces all error conditions, as well as process and device initialization messages.
  • Special—Traces all special conditions and subsystem state transitions that occur during normal operation. Traces call-processing events.
• **State Transition**—Traces all state transition conditions and media layer events that occur during normal operation.

• **Significant**—Traces all significant conditions, as well as entry and exit points of routines. Not all services use this trace level.

• **Entry_exit**—Traces all entry and exit conditions, plus low-level debugging information.

• **Arbitrary**—Traces all Arbitrary conditions plus detailed debugging information.

• **Detailed**—Traces alarm conditions and events. Used for all traces that are generated in abnormal path. Uses minimum number of CPU cycles.

The default value is **Error**.

**Tip** We recommend that you check all the check boxes in this section for troubleshooting purposes.

**Step 7** Click **Save**.

---

**What to do next**

(Optional) **Configure the Service Parameters for the Quality Report Tool, on page 355**

**Configure the Service Parameters for the Quality Report Tool**

---

**Caution**

We recommend that you use the default service parameters settings unless the Cisco Technical Assistance Center (TAC) instructs otherwise.

---

**Procedure**

---

**Step 1** From Cisco Unified Communications Manager Administration, choose **System > Service Parameters**.

**Step 2** Choose the node where the QRT application resides.

**Step 3** Choose the **Cisco Extended Functions** service.

**Step 4** Configure the service parameters. See the Related Topics section for more information about the service parameters and their configuration options.

**Step 5** Click **Save**.

---

**Related Topics**

[Quality Report Tool Service Parameters, on page 356](#)
### Quality Report Tool Service Parameters

Table 58: Quality Report Tool Service Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| Display Extended QRT Menu Choices  | Determines whether extended menu choices are presented to the user. You can choose one of the following configuration options:  
  • Set this field to true to display extended menu choices (interview mode).  
  • Set this field to false to not display extended menu choices (silent mode).  
  • The recommended default value is false (silent mode).                                                                                                                                  |
| Streaming Statistics Polling Duration | Determines the duration that is to be used for polling streaming statistics. You can choose one of the following configuration options:  
  • Set this field to -1 to poll until the call ends.  
  • Set this field to 0 to not poll at all.  
  • Set it to any positive value to poll for that many seconds. Polling stops when the call ends.  
  • The recommended default value is -1 (poll until the call ends).                                                                                                                               |
| Streaming Statistics Polling Frequency (seconds) | Enter the number of seconds to wait between each poll.  
  The value ranges between 30 and 3600. The recommended default value is 30.                                                                                                     |
| Maximum No. of Files                | Enter the maximum number of files before the file count restarts and overwrites the old files.  
  Valid values are between 1 and 10000. The recommended default value is 250.                                                                                                         |
| Maximum No. of Lines per File       | Enter the maximum number of lines in each file before starting the next file:  
  • The value ranges between 100 and 2000.  
  • The recommended default value specifies 2000.                                                                                                                                       |
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
<th>Note</th>
</tr>
</thead>
<tbody>
<tr>
<td>CAPF Profile Instance Id for Secure Connection to CTI Manager</td>
<td>Enter the Instance ID of the Application CAPF Profile for application user CCMQRTSysUser that the Cisco Extended Function service will use to open a secure connection to CTI Manager. You must configure this parameter if CTI Manager Connection Security Flag is enabled.</td>
<td>Turn on security by enabling the CTI Manager Connection Security Flag service parameter. You must restart the Cisco Extended Functions service for the changes to take effect.</td>
</tr>
<tr>
<td>CTI Manager Connection Security Flag</td>
<td>Choose whether security for Cisco Extended Functions service CTI Manager connection is enabled or disabled. If enabled, Cisco Extended Functions will open a secure connection to CTI Manager using the Application CAPF Profile configured for the Instance ID for application user CCMQRTSysUser. The value choices are True and False. You must choose True to enable a secure connection to CTI.</td>
<td></td>
</tr>
</tbody>
</table>
Third-Party SIP Endpoints Overview

In addition to the Cisco IP Phones that run SIP, Unified Communications Manager supports a variety of third-party SIP endpoints. You can configure the following third-party SIP endpoints in Cisco Unified Communications Manager Administration:

- Third-Party SIP Device (Advanced)—This eight-line SIP device is an RFC3261-compliant phone that is running SIP from third-party companies.
- Third-Party SIP Device (Basic)—This one-line SIP device is an RFC3261-compliant phone that is running SIP from third-party companies.
- Third-Party AS-SIP Device—Assured Services SIP (AS-SIP) endpoints are SIP endpoints compliant with MLPP, DSCP, TLS/SRTP, and IPv6 requirements. AS-SIP provides multiple endpoint interfaces on the Unified Communications Manager.
- Generic Desktop Video Endpoint—This SIP device supports video, security, configurable trust, and Cisco extensions. This device supports 8 lines; the maximum number of calls and busy trigger for each line is 4 and 2, respectively.
- Generic Single Screen Room System—This SIP device supports single screen telepresence (room systems), video, security, configurable trust, and Cisco extensions. This device supports 8 lines; the maximum number of calls and busy trigger for each line is 4 and 2, respectively.
- Generic Multiple Screen Room System—This SIP device supports multiple screen telepresence (room systems), video, security, configurable trust, and Cisco extensions. This device supports 8 lines; the maximum number of calls and busy trigger for each line is 4 and 2, respectively.

Third-Party SIP Endpoints Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure a Digest User, on page 360</td>
<td>To enable digest authentication, configure an end user that is a digest user. Cisco Unified Communications Manager Administration.</td>
</tr>
</tbody>
</table>
Configure a Digest User

Perform these steps to configure an end user as a digest user. Digest authentication allows Cisco Unified Communications Manager to challenge the identity of a device that is connecting to Cisco Unified Communications Manager. When challenged, the device presents its digest credentials, similar to a username and password, to Cisco Unified Communications Manager for verification. If the credentials that are presented match those that are configured in the database for that device, digest authentication succeeds, and Cisco Unified Communications Manager processes the SIP request.

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose User Management > End User.

**Step 2**  
Click Add New.

**Step 3**  
Enter a User ID.

**Step 4**  
Enter a Last Name.

**Communications Manager** uses the digest credentials that you specify in the End User Configuration window to validate the SIP user agent response during a challenge to the SIP trunk.

If the third-party SIP phone does not support a digest user, create a user with a user ID that matches the directory number of the third-party SIP phone. For example, create an end user named 1000 and create a directory number of 1000 for the phone. Assign this user to the phone.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configure SIP Profile, on page 324</td>
<td>Provide a set of SIP attributes that are associated with SIP trunks.</td>
</tr>
<tr>
<td>Configure Phone Security Profile, on page 324</td>
<td>To use digest authentication, you must configure a new phone security profile. If you use one of the standard, nonsecure SIP profiles that are provided for auto-registration, you cannot enable digest authentication.</td>
</tr>
<tr>
<td>Add a Third-Party SIP Endpoint, on page 362</td>
<td>Configure a third-party endpoint.</td>
</tr>
<tr>
<td>Associate Device to End User, on page 363</td>
<td>Associate the third-party endpoint with an end user.</td>
</tr>
</tbody>
</table>

**What to do next**

Provide power, verify network connectivity, and configure network settings for the third-party SIP endpoint. For more information about configuring network settings, see the user guide for your third-party SIP endpoint.
Configure Endpoint Devices

Configure SIP Profile

Use this procedure to configure SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks.

**Before you begin**
- Configure SIP Phone Secure Port, on page 323
- Restart Services, on page 323

**Procedure**

1. In Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.
2. Click Find.
3. For the profile that you want to copy, click the file icon in the Copy column.
4. Enter the name and description of the new profile.
5. If you have the IPv6 stack configured and you are deploying two stacks, check the Enable ANAT check box.
   **Note** This configuration applies whether you have Unity Connection deployed or not.
6. Click Save.

**What to do next**
Configure Phone Security Profile, on page 324

Configure Phone Security Profile

Cisco Unified Communications Manager provides a set of predefined, nonsecure profiles for autoregistration. If you want to enable security features for a phone, you must configure a new security profile and apply it to the phone. Follow these steps to configure a new security profile:

**Before you begin**
If you are configuring SIP phones, complete the following procedures:
Add a Third-Party SIP Endpoint

Before you begin
Configure a Digest User, on page 360

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Perform one of the following steps:
  • Click Add New to create a new third-party endpoint.
  • Click Find to search and select an existing third-party endpoint.
Step 3 From the Phone Type drop-down list, choose one of the following:
  • Third-party SIP Device (Basic)
  • Third-party SIP Device (Advanced)
Configure Endpoint Devices

- Third-Party AS-SIP Device
- Third-party AS-SIP Endpoint
- Generic Desktop Video Endpoint
- Generic Single Screen Room System
- Generic Multiple Screen Room System

Step 4  From the **Digest User** drop-down list, choose the user that you created.

Step 5  Configure the fields in the **Phone Configuration** window. See the online help for more information about the fields and their configuration options.

Step 6  Click **Save**.

Step 7  To configure a directory number for the third-party endpoint, click the **Add a New DN** link that displays in the **Association Information** area on the left side of the window. The **Directory Number Configuration** window appears.

Step 8  Configure the fields in the Directory Number Configuration window. For help with the fields and their settings, see the online help.

---

What to do next

**Associate Device to End User, on page 363**

## Associate Device to End User

Use this procedure to associate an end user to the third-party endpoint.

### Procedure

**Step 1**  From Cisco Unified CM Administration, choose **User Management > End User**.

**Step 2**  Click **Find** and select the user whom you want to associate to the device.

**Step 3**  In the **Device Information** section, click **Device Association**. The User Device Association window appears.

**Step 4**  Click **Find** to view a list of available devices.

**Step 5**  Select the device that you want to associate, and click **Save Selected/Changes**.

**Step 6**  From **Related Links**, choose **Back to User**, and click **Go**. The **End User Configuration** window appears, and the associated device that you chose appears in the **Controlled Devices** pane.
## Third-Party Interactions and Restrictions

### Third-Party Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ringback tone restriction for Cisco Video Communications Server (VCS) registered to third-party SIP Endpoints</td>
<td>Blind transfer or switch to request the transfer which occurs over VCS registered endpoints with Cisco Unified Communications Manager will not have a ringback tone. If you do a supervised transfer, then you allocate Music On Hold (MOH) but, not a ringback tone.</td>
</tr>
</tbody>
</table>
Device Profiles and Templates

- Device Profiles and Templates Overview, on page 365
- Configure Device Profiles and Templates Task Flow, on page 366

Device Profiles and Templates Overview


Device Profiles

A device profile defines the services, features, and directory numbers that associate with a particular device. You can configure a device profile and then you can assign the user device profile to a user, so that when the user logs in to a device, those features and services are available on that device.

SIP Profiles for End Points

A SIP profile comprises the set of SIP attributes that are associated with SIP endpoints. SIP profiles include information such as name, description, timing, retry, call pickup URI, and so on. The profiles contain some standard entries that cannot be deleted or changed.

Device Profiles and Templates

Cisco Unified Communications Manager also supports a default device profile. Cisco Unified Communications Manager uses the default device profile whenever a user logs on to a phone model for which no user device profile exists.

Peer-to-Peer Image Distribution

The peer firmware sharing feature provides these advantages in high-speed campus LAN settings:

- Limits congestion on TFTP transfers to centralized TFTP servers.
- Eliminates the need to manually control firmware upgrades.
• Reduces phone downtime during upgrades when large numbers of devices are reset simultaneously.

In most conditions, the peer firmware sharing feature optimizes firmware upgrades in branch deployment scenarios over bandwidth-limited WAN links.

When the feature is enabled, it allows the phone to discover similar phones on the subnet that are requesting the files that make up the firmware image and to automatically assemble transfer hierarchies on a per-file basis. The individual files that make up the firmware image get retrieved from the TFTP server by only the root phone in the hierarchy and are then rapidly transferred down the transfer hierarchy to the other phones on the subnet using TCP connections.

## Configure Device Profiles and Templates Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure a Softkey Template on the Default Device Profile, on page 367</td>
<td>Add the default device profile to a softkey template.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Softkey Template with a Common Device Configuration, on page 368</td>
<td>Optional. To make the softkey template available to phones, you must associate the template to a Common Device Configuration or directly to a phone. Follow this step if your system uses a common device configuration to apply configuration options to phones (this is the most commonly used method for making a softkey template available to phones).</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Associate a Softkey Template with a Phone, on page 370</td>
<td>Optional. Use this procedure either as an alternative to associating the softkey template with the common device configuration, or in conjunction with the common device configuration. Use this procedure in conjunction with the common device configuration if you need assign a softkey template that overrides the assignment in the common device configuration or any other default softkey assignment.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure Feature Control Policy Task Flow, on page 370</td>
<td>Optional. Use this procedure as an alternative to configuring softkey templates. Configure</td>
</tr>
</tbody>
</table>

Configure a Softkey Template on the Default Device Profile

Cisco Unified Communications Manager includes standard softkey templates for call processing and applications. When creating custom softkey templates, copy the standard templates and make modifications as required.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose Device > Device Settings > Softkey Template.

**Step 2**
Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click Add New.
b) Select a default template and click Copy.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click Save.

**Step 3**
Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

**Step 4** Check the Default Softkey Template check box to designate this softkey template as the default softkey template.

*Note* If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

**Step 5** Choose Configure Softkey Layout from the Related Links drop-down list in the upper right corner and click Go.

**Step 6** From the Select a Call State to Configure drop-down list, choose the call state for which you want the softkey to display.

**Step 7** From the Unselected Softkeys list, choose the softkey to add and click the right arrow to move the softkey to the Selected Softkeys list. Use the up and down arrows to change the position of the new softkey.

**Step 8** Repeat the previous step to display the softkey in additional call states.

**Step 9** Click Save.

**Step 10** Perform one of the following tasks:

- Click Apply Config if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see Add a Softkey Template to a Common Device Configuration and Associate a Softkey Template with a Phone sections.

---

**What to do next**

You can apply a customized softkey template to a device by selecting the template from the Softkey Template drop-down in one of the following configuration windows:

- Phone Configuration
- Universal Device Template
- BAT Template
- Common Device Configuration
- Device Profile
- Default Device Profile
- UDP Profile

---

**Associate a Softkey Template with a Common Device Configuration**

*Optional.* There are two ways to associate a softkey template with a phone:

- Add the softkey template to the Phone Configuration.
- Add the softkey template to the Common Device Configuration.
The procedures in this section describe how to associate the softkey template with a Common Device Configuration. Follow these procedures if your system uses a Common Device Configuration to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.

To use the alternative method, see the section Associate a Softkey Template with a Phone.

Procedure

Add a Softkey Template to a Common Device Configuration

Procedure

Step 1  Add a Softkey Template to a Common Device Configuration, on page 369
Step 2  Associate a Common Device Configuration with a Phone, on page 369

Add a Softkey Template to a Common Device Configuration

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Device Settings > Common Device Configuration.
Step 2  Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.
   a)  Click Add New.
   b)  Enter a name for the Common Device Configuration in the Name field.
   c)  Click Save.
Step 3  Perform the following steps to add the softkey template to an existing Common Device Configuration.
   a)  Click Find and enter the search criteria.
   b)  Click an existing Common Device Configuration.
Step 4  In the Softkey Template drop-down list, choose the softkey template that contains the softkey that you want to make available.
Step 5  Click Save.
Step 6  Perform one of the following tasks:
   •  If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.
   •  If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Click Find and select the phone device to add the softkey template.
**Associate a Softkey Template with a Phone**

*Optional.* Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > Phone.

**Step 2** Click Find to select the phone to add the softkey template.

**Step 3** From the Softkey Template drop-down list, choose the template that contains the new softkey.

**Step 4** Click Save.

**Step 5** Press Reset to update the phone settings.

**Configure Feature Control Policy Task Flow**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Generate a Phone Feature List, on page 371</td>
<td>Login to Cisco Unified Reporting and run a phone feature list report to determine which phones support Feature Control Policy.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Create a Feature Control Policy, on page 371</td>
<td>Create a Feature Control Policy for Cisco IP Phones.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Perform one of the following tasks:</td>
<td>After you create a Feature Control Policy, you must associate that policy to an individual phone, a group of phones, or to all phones in the system. The Feature Control Policy for an individual phone overrides the clusterwide feature control policy.</td>
</tr>
<tr>
<td>• Apply Feature Control Policy to a Phone, on page 372</td>
<td></td>
</tr>
<tr>
<td>• Apply Feature Control Policy to a Common Phone Profile, on page 372</td>
<td></td>
</tr>
<tr>
<td>• Apply Feature Control Policy to All Phones, on page 373</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>For information on how to apply Feature Control Policy on multiple phones by using the Bulk Administration Tool, see the <em>Cisco Unified Communications Manager Bulk Administration Guide</em>.</td>
</tr>
</tbody>
</table>

### Generate a Phone Feature List

Generate a phone feature list report to determine which devices support the feature that you want to configure.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **System Reports**.

**Step 2** From the list of reports, click **Unified CM Phone Feature List**.

**Step 3** Perform one of the following steps:

- Choose **Generate New Report** (the bar chart icon) to generate a new report.
- Choose **Unified CM Phone Feature List** if a report exists.

**Step 4** From the **Product** drop-down list, choose **All**.

**Step 5** Click the name of the feature that you want to configure.

**Step 6** Click **Submit**, to generate the report.

### Create a Feature Control Policy

Follow these steps to create a feature control policy. Use this policy to enable or disable a particular feature and hence control the appearance of softkeys that display on the phone.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Device Settings > Feature Control Policy**.

**Step 2** Perform one of the following tasks:

- To modify the settings for an existing policy, enter search criteria, click **Find** and choose the policy from the resulting list.

- To add a new policy, click **Add New**.

The **Feature Control Policy Configuration** window is displayed.

**Step 3** In the **Name** field, enter a name for the feature control policy.

**Step 4** In the **Description** field, enter a brief description for the feature control policy.

**Step 5** In the **Feature Control Section**, for each feature listed, choose whether you want to override the system default and enable or disable the setting:
Apply Feature Control Policy to a Phone

Before you begin

• Ensure that the phone model supports Feature Control Policy. For more information, see Generate a Phone Feature List, on page 371.

• Create a Feature Control Policy, on page 371

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Enter the search criteria and click Find. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.
Step 3 Choose a phone to which you want to apply a Feature Control Policy.
Step 4 From the Feature Control Policy drop-down list, choose the required Feature Control Policy.
Step 5 Click Save.
Step 6 Click Apply Config.
Step 7 Click OK.

Apply Feature Control Policy to a Common Phone Profile

Common Phone Profiles allow you to configure Feature Control Policy settings and then apply those settings to all the phones in your network that use that profile.

Before you begin

Create a Feature Control Policy, on page 371

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Common Phone Profile.
Step 2 Enter the search criteria and click Find.
Step 3 Choose a common phone profile to which you want to apply a Feature Control Policy.
Step 4 From the Feature Control Policy drop-down list, choose the required Feature Control Policy.
Step 5  Click Save.
Step 6  Click Apply Config.
Step 7  Click OK.

**Apply Feature Control Policy to All Phones**

**Before you begin**

Create a Feature Control Policy, on page 371

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose **System > Enterprise Parameters**.
**Step 2**  From the **Feature Control Policy** drop-down list, choose the required Feature Control Policy.
**Step 3**  Click Save.
**Step 4**  Click Apply Config.
**Step 5**  Click OK.

**Configure a Phone Button Template**

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose **Device > Device Settings > Phone Button Template**.
**Step 2**  Click Find to display list of supported phone templates.
**Step 3**  Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.
   a)  Select a default template for the model of phone and click Copy.
   b)  In the **Phone Button Template Information** field, enter a new name for the template.
   c)  Click Save.
**Step 4**  Perform the following steps if you want to add phone buttons to an existing template.
   a)  Click Find and enter the search criteria.
   b)  Choose an existing template.
**Step 5**  From the **Line** drop-down list, choose feature that you want to add to the template.
**Step 6**  Click Save.
**Step 7**  Perform one of the following tasks:
   - Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
• If you created a new softkey template, associate the template with the devices and then restart them.

## Associate a Button Template with a Phone

**Before you begin**

Configure a Phone Button Template, on page 373

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > Phone.

**Step 2** Click Find to display the list of configured phones.

**Step 3** Choose the phone to which you want to add the phone button template.

**Step 4** In the Phone Button Template drop-down list, choose the phone button template that contains the new feature button.

**Step 5** Click Save.

A dialog box is displayed with a message to press Reset to update the phone settings.

## Configure Device Profile

A device profile comprises the set of attributes that associate with a particular device. You can associate the device profile that you create to an end user in order to use the Cisco Extension Mobility feature.

**Procedure**

**Step 1** From the Cisco Unified CM Administration window, choose Device > Device Settings > Device Profile.

**Step 2** In the Device Profile Configuration window, from the Device Profile Type drop-down list, choose the appropriate Cisco Unified IP Phone.

**Step 3** Click Next.

**Step 4** From the Device Protocol drop-down list, choose the appropriate protocol.

**Step 5** Click Next.

**Step 6** From the Phone Button Template drop-down list, choose a template.

**Step 7** (Optional) From the Softkey Template drop-down list, select a softkey template.

**Step 8** Configure the fields in the Device Profile Configuration window. See the online help for more information about the fields and their configuration options.

**Step 9** Click Save.

**Note** For details on using Device Profiles to setup Cisco Extension Mobility, see the Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1.
Configure SIP Profiles for Endpoints

Cisco Unified Communications Manager uses SIP profiles to define SIP attributes that are associated with SIP trunks and Cisco Unified IP Phones.

Procedure

Step 1  In the Cisco Unified CM Administration window, choose Device > Device Settings > SIP Profiles.
Step 2  To add a new SIP Profile, click the Add New button.
Step 3  Configure the fields in the SIP Profile Configuration window. See the online help for more information about the fields and their configuration options.
Step 4  Click Apply Config.

Configure Default Device Profiles

The phone takes on the default device profile whenever a user logs into a phone for which that user does not have a user device profile.

A default device profile includes device type (phone), user locale, phone button template, softkey template, and multilevel precedence and preemption (MLPP) information.

Procedure

Step 1  From the Cisco Unified CM Administration window, choose Device > Device Settings > Default Device Profile.
Step 2  In the Default Device Profile Configuration window, from the Device Profile Type drop-down list, choose the appropriate Cisco Unified IP Phone.
Step 3  Click Next.
Step 4  From the Device Protocol drop-down list, choose the appropriate protocol.
Step 5  Click Next.
Step 6  Configure the fields in the Default Device Profile Configuration window. See the online help for more information about the fields and their configuration options.
Step 7  Click Save.

Configure Peer-to-Peer Image Distribution Feature for Phones

Procedure

Step 1  In Cisco Unified Communications Manager Administration, choose Device > Phone.
Step 2  In the **Find and List Phones** window, to select a phone, specify the appropriate filters in the **Find Phone where** field, click **Find** to retrieve a list of phones, and then select the phone from the list.

Step 3  In the **Phone Configuration** window, from the **Peer Firmware Sharing** drop-down list in the Product Specific Configuration Layout pane, choose one the following options.

- **Enabled** (default)—Indicates that the phone supports Peer-to-Peer Image Distribution (PPID).
- **Disabled**—Indicates that the phone does not support Peer-to-Peer Image Distribution (PPID).

Step 4  Click **Apply Config**.
Associate Users with Endpoints

• Users to Endpoints Association Overview, on page 377
• Associate Users with Endpoints Prerequisites, on page 377
• Users and Devices Configuration Task Flow, on page 377
• Interactions and Restrictions for Associating Users with Endpoints, on page 381

Users to Endpoints Association Overview

This chapter describes how to associate devices with end users and application users. End users can control the devices that you associate with them. Applications that are identified as users can control devices, such as phones and Computer Telephony Integration (CTI) ports.

Associate Users with Endpoints Prerequisites

Configure end users and application users before you associate them with endpoints. See Associate End Users with Devices, on page 377 and Associate Application Users with Devices, on page 380.

Users and Devices Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Associate End Users with Devices, on page 377</td>
<td>Associate end users with devices.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate Application Users with Devices, on page 380</td>
<td>Associate application users with devices.</td>
</tr>
</tbody>
</table>

Associate End Users with Devices

Cisco Unified Communications Manager does not allow duplicate end user IDs.
**Procedure**

**Step 1** From Cisco Unified CM Administration, choose User Management > End User.

**Step 2** From the Find and List Application Users window, click Find.

**Step 3** From the window that displays a list of end users, click the link for the relevant end user.

**Step 4** From the End User Configuration window, scroll down to the Device Information area and select the devices that you want to associate with the end user. In the Available Devices box, choose a device that you want to associate with the application user and click the down arrow below the box.

**Note** If no devices exist in the Device Information area, click the Device Association button to open the User Device Association window. Select one or multiple devices and click the Save Selected/Changes button. The selected devices appear in the Controlled Devices list box of the Device Information area. Then, follow Steps 1 to 4 and associate a device.

**Step 5** (Optional) To associate a line appearance to an end user for presence and to enable the on-the-phone status information to IM and Presence clients when this line appearance is off-hook, click the Line Appearance Association from Presence button. The Line Appearance Association for Presence window appears from where you can choose product type, device name, directory, partition, or description. The choices available in this window depend on the lines associated with the controlled devices. Click Save.

**Step 6** Configure the fields from the End User Configuration window. See the Related Topics section for more information about the fields and their configuration options.

**Step 7** Click Save.

**What to do next**

Associate Application Users with Devices, on page 380.

**Related Topics**

End User and Device Configuration Settings, on page 378

### End User and Device Configuration Settings

#### Table 60: User Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>User ID</td>
<td>Enter the end user identification name. Cisco Unified Communications Manager does not permit modifying the user ID after it is created. You may use the following special characters: =, +, &lt;, &gt;, #, :, , &quot;, “, and blank spaces.</td>
</tr>
<tr>
<td>Password</td>
<td>Enter five or more alphanumeric or special characters for the end user password. You may use the following special characters: =, +, &lt;, &gt;, #, :, , “, “, and blank spaces</td>
</tr>
<tr>
<td>PIN</td>
<td>Enter five or more numeric characters for the Personal Identification Number.</td>
</tr>
</tbody>
</table>
### End User and Device Configuration Settings

**Table 61: Device Associations**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Last Name</td>
<td>Enter the end user last name. You may use the following special characters: =,+,&lt;,&gt;,#,;,,&quot;,&quot;, and blank spaces</td>
</tr>
<tr>
<td>Middle Name</td>
<td>Enter the end user middle name. You may use the following special characters: =,+,&lt;,&gt;,#,;,,&quot;,&quot;, and blank spaces</td>
</tr>
<tr>
<td>First Name</td>
<td>Enter the end user first name. You may use the following special characters: =,+,&lt;,&gt;,#,;,,&quot;,&quot;, and blank spaces</td>
</tr>
<tr>
<td>Product Type</td>
<td>From the drop-down list, choose the type of device to associate with this end user.</td>
</tr>
<tr>
<td>MAC Address</td>
<td>Enter a unique MAC address for the new device that you are associating with the new user. The MAC address comprises exactly 12 hexadecimal digits (0 to 9, A to F).</td>
</tr>
<tr>
<td>Calling Search Space DN</td>
<td>From the drop-down list, choose the calling search space for the directory number that you are associating with this user and device.</td>
</tr>
<tr>
<td>Calling Search Space Phone</td>
<td>From the drop-down list, choose the calling search space for the phone that you are associating with this user and device.</td>
</tr>
</tbody>
</table>
### Associate Application Users with Devices

You can associate devices over which application users have control. Application users can control devices, such as phones. Applications that are identified as users can control other devices, such as CTI ports. When application users have control of a phone, they can control certain settings for that phone, such as speed dial and call forwarding.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>External Phone Number Mask</strong></td>
<td>Specify the mask that is used to format caller ID information for external (outbound) calls that are made from the associated device.  &lt;br&gt;• The mask can contain up to 24 characters. Valid characters specify 0 to 9, *, #, and X.  &lt;br&gt;• Enter the literal digits that you want to appear in the caller ID information and use Xs to represent the directory number of the associated device.  &lt;br&gt;• If you specify a mask of 972813xxxx, an external call from extension 1234 displays a caller ID number of 9728131234 if the Use External PhoneNumber Mask option is checked on the route pattern that is used to make the external call. If you specify a mask of all literal digits, such as 9728135000 to represent a main attendant number, that literal number (9728135000) displays as the caller ID for an external call from any associated device.</td>
</tr>
<tr>
<td><strong>Extension</strong></td>
<td>Enter an extension for the new user and phone. You may use the following characters: 0 to 9, ?, [ ], +, -, *, ^, #, !.  &lt;br&gt;This field represents the primary directory number for the end user. End users can have multiple lines on their phones.</td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
<td>From the drop-down list, choose a partition for the directory number that you specified in the Extension field.</td>
</tr>
<tr>
<td><strong>Voice Mail Profile</strong></td>
<td>From the drop-down list, choose a voice mail profile for the directory number.  &lt;br&gt;Choose None to use the system default.</td>
</tr>
<tr>
<td><strong>Enable Extension Mobility</strong></td>
<td>Check this check box to enable extension mobility.  &lt;br&gt;After you add the new user, you can use the <strong>User Management &gt; End User</strong> menu option to choose an Extension Mobility profile.</td>
</tr>
</tbody>
</table>
Before you begin

Associate End Users with Devices, on page 377.

Procedure

**Step 1** From Cisco Unified CM Administration, choose User Management > Application User. appears.

**Step 2** From the Find and List Application Users window, click Find.

**Step 3** From the list of application users, click the link for the relevant Application User.

**Step 4** From the Application User Configuration window, scroll down to the Device Information area. In the Available Devices box, choose a device that you want to associate with the application user and click the down arrow below the box.

The device is moved to the Controlled Devices box.

**Step 5** To add to the list of available devices, click one of the following buttons:

- **Find more Phones**—To find the phones to associate with this application user.
- **Find more Route Points**—To find the CTI route points to associate with this application user.
- **Find more Pilot Points**—To find the pilot points to associate with this application user.

**Step 6** Repeat Step 5 for each device that you want to assign to the application user.

**Step 7** Click Save.

---

Interactions and Restrictions for Associating Users with Endpoints

**Interactions for Associating Users with Endpoints**

<table>
<thead>
<tr>
<th>Table 62: Users with Endpoints Association Interactions</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Feature</strong></td>
</tr>
<tr>
<td>Non-CTI-controllable devices</td>
</tr>
<tr>
<td>Cisco Extension Mobility</td>
</tr>
<tr>
<td>IM and Presence Service</td>
</tr>
</tbody>
</table>
# Restrictions for Associating Users with Endpoints

## Table 63: Users with Endpoints Association Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Modification of end user information</td>
<td>You can modify end user information only if synchronization with an LDAP server is not enabled. To check whether synchronization with an LDAP server is enabled, choose <strong>System &gt; LDAP &gt; LDAP System</strong>.</td>
</tr>
</tbody>
</table>
PART VII

Integrate Applications

- Integrate Applications Overview, on page 385
- Configure Application Servers, on page 387
- Install Plugins, on page 391
- Configure Presence Redundancy Groups, on page 395
- Configure Cisco Unity Connection for Voicemail and Messaging, on page 411
- Configure Cisco Unified Contact Center Enterprise, on page 415
- Configure Cisco Unified Contact Center Express, on page 417
- Configure CTI Applications, on page 419
- Configure Cisco TelePresence, on page 429
- Configure Cisco Jabber, on page 431
Integrate Applications Overview

- About Integrating Applications, on page 385
- Integrate Applications, on page 385

About Integrating Applications

The chapters in this part describe how to extend the functionality of your system by integrating applications. You can add various functions such as voicemail, contact center features, rich conferencing, or features to monitor the health of your system. Some applications, such as the Cisco Unified Real-Time Monitoring Tool, are built into your system and can be downloaded from the administration interface. Other applications, such as Cisco Jabber or Cisco Unified Contact Center Express, are external to your system and can be configured to interoperate with Unified Communications Manager.

Integrate Applications

Complete the following task flows to integrate applications for your system.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Application Servers Task Flow, on page 387</td>
<td>Configure application servers to add other product servers to your cluster and establish secure operations between them.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Install Plugins Task Flow, on page 392</td>
<td>Use application plugins to extend the functionality of your system.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Presence Redundancy Group Task Flow, on page 396</td>
<td>Configure a presence redundancy group which is comprised of two IM and Presence Service nodes from the same cluster. This group provides both redundancy and recovery for IM and Presence Service clients and applications.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Cisco Unity Connection, on page 411</td>
<td>Integrate Cisco Unity Connection with your system so that you can provide voicemail and messaging features to your users.</td>
</tr>
<tr>
<td>Step</td>
<td>Command or Action</td>
</tr>
<tr>
<td>------</td>
<td>------------------</td>
</tr>
<tr>
<td>5</td>
<td>Cisco Unified Contact Center Enterprise, on page 415</td>
</tr>
<tr>
<td>6</td>
<td>Cisco Unified Contact Center Express, on page 417</td>
</tr>
<tr>
<td>7</td>
<td>Configure CTI Applications Task Flow, on page 421</td>
</tr>
<tr>
<td>8</td>
<td>Cisco TelePresence, on page 429</td>
</tr>
<tr>
<td>9</td>
<td>Configure Cisco Jabber, on page 431</td>
</tr>
</tbody>
</table>
CHAPTER 48

Configure Application Servers

- Application Servers Overview, on page 387
- Application Servers Prerequisites, on page 387
- Application Servers Task Flow, on page 387

Application Servers Overview

Use the application server function to maintain associations between the Cisco Unified Communications Manager and off-cluster, external applications, such as Cisco Unity Connection and Cisco Emergency Responder. Application servers also synchronize information between Cisco Unified Communications Manager and applications such as Cisco WebDialer.

Application Servers Prerequisites

For Cisco Unity and Cisco Unity Connection, make sure that the AXL web service is running on the Cisco Unified Communications Manager node that is configured to communicate with the Cisco Unity and Cisco Unity Connection server.

Application Servers Task Flow

Perform either of the following tasks, depending on the type of application server that you want to configure.

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Configure Application Servers, on page 388</td>
<td>Configure application servers that you want to securely join, interoperate, and share information within your cluster.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Cisco WebDialer Servers, on page 388</td>
<td>Configure Cisco WebDialer application servers as an alternative to the List of WebDialers service parameter, which limits the number of characters that you can enter. After you add a Cisco WebDialer application server in the</td>
</tr>
</tbody>
</table>
Configure Application Servers

Configure application servers that you want to securely join, interoperate, and share information within your cluster.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose System &gt; Application Server.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Application Server Type drop-down list, choose one of the following server options:</td>
</tr>
<tr>
<td></td>
<td>• Cisco Unity Voice Mail 4.x or later</td>
</tr>
<tr>
<td></td>
<td>• Cisco Unity Connection</td>
</tr>
<tr>
<td></td>
<td>• CUMA Provisioning Server</td>
</tr>
<tr>
<td></td>
<td>• CER Location Management</td>
</tr>
<tr>
<td></td>
<td>• Remote System Log Server</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Next.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure the fields on the Application Server Configuration window. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Configure Cisco WebDialer Servers

Procedure

**Step 1**
From Cisco Unified CM Administration, choose **System > Application Server**.

**Step 2**
Click **Add New**.

**Step 3**
From the **Application Server Type** drop-down list, choose **Cisco Web Dialer**, and then click **Next**.

**Step 4**
In the **Hostname or IP Address** field, enter the hostname or IP address of the WebDialer server.

**Step 5**
From the **Redirector Node** drop-down list, choose **< None >** or a specific Unified Communications Manager node.

**< None >** indicates the WebDialer Server would apply to all nodes.

**Step 6**
Click **Save**.

**Step 7**
From Cisco Unified Serviceability, choose **Tools > Control Center - Feature Services**

**Step 8**
Click the **Cisco WebDialer Web Service** radio button.

**Step 9**
Click **Restart**.
CHAPTER 49

Install Plugins

• Plugins Overview, on page 391
• Install Plugins Task Flow, on page 392

Plugins Overview

Application plugins extend the functionality of your system.

The following plugins are available from the Application > Plugins menu:

• Cisco AXL Toolkit—Lets developers create applications that create, read, update and delete provisioning objects on the publisher node. The zip file contains Java-based libraries that use SOAP over HTTP/HTTPS to send and receive AXL requests and responses.

• Cisco CTL Client—Digitally signs certificates that are stored on the TFTP server. The client retrieves the CTL file from the Cisco TFTP server, digitally signs the CTL file using a security token and then updates the file on the Cisco TFTP server.

• Cisco IP Phone Address Book Synchronizer—Synchronizes a Microsoft Windows address book with a Cisco personal address book.

• Cisco JTAPI Client—Provides a standard programming interface for communication-enabled applications that are written in the Java programming language.

• Cisco TAPI Client—Provides a standard programming interface for communication-enabled applications that are running on Microsoft Windows.

• Cisco Tool for Auto-Registered Phone Support (TAPS)—Helps users remotely download preconfigured phone settings to provision their devices.

• Cisco Unified CM Assistant Console—Helps assistants more effectively handle calls for their managers. The assistant console connects to the Cisco Unified Communications Manager IP Manager Assistant (IPMA) Service for login and directory services.

• Cisco Unified Real-Time Monitoring Tool—Monitors device status system performance device discovery and CTI applications running on your cluster in real-time. RTMT also connects directly to devices to aid in troubleshooting.
Install Plugins Task Flow

Perform the following tasks as needed.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Download a Plugin, on page 392</td>
<td>Download a plugin, and then follow installation instructions from the executable or ZIP file. After you upgrade your system, you must reinstall all plugins.</td>
</tr>
<tr>
<td>Step 2</td>
<td>(Optional) Update the Plugin URLs, on page 393</td>
<td>Update the plugin URLs if your domain name server (DNS) changes. At the time of your system installation, the DNS provides the basis for the plugin URL. If the DNS changes, the URL is not automatically updated.</td>
</tr>
</tbody>
</table>

Download a Plugin

Download a plugin, and then follow installation instructions from the executable or ZIP file. After you upgrade your system, you must reinstall all plugins.

**Before you begin**

Temporarily disable all intrusion detection or antivirus services that run on the server where you plan to install the plugin.

**Procedure**

1. From Cisco Unified CM Administration, choose Application > Plugins.
2. Enter search criteria or leave the dialog box blank, and then click Find.
   - The window that appears contains more information about the application plugins.
3. Click Download for the plugin that you want to download and install.
   - You can also right click Download and click Save As to choose a folder that is easy for you to find.
4. (Optional) If your plugin is a ZIP file, unzip this file using a built-in or third-party zip program.
5. Run the executable file or, if applicable, consult the readme file contained in a ZIP file.

**What to do next**

Walk through the instructions in the executable file to install the plugin.
Update the Plugin URLs

Update the plugin URLs if your domain name server (DNS) changes. At the time of your system installation, the DNS provides the basis for the plugin URL. If the DNS changes, the URL is not automatically updated.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Application</strong> &gt; <strong>Plugins</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click the plugin name that you want to update.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Custom URL</strong> field, enter the updated URL for the plugin.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>
Update the Plugin URLs
CHAPTER 50

Configure Presence Redundancy Groups

• Presence Redundancy Group Overview, on page 395
• Presence Redundancy Group Prerequisites, on page 396
• Presence Redundancy Group Task Flow, on page 396
• Redundancy Interactions and Restrictions, on page 401
• Initiate Manual Failover, Fallback, or Recovery, on page 402

Presence Redundancy Group Overview

A presence redundancy group is comprised of two IM and Presence Service nodes from the same cluster. Each node in the presence redundancy group monitors the status, or heartbeat, of the peer node. You can configure a presence redundancy group to provide both redundancy and recovery for IM and Presence Service clients and applications.

• Failover—Occurs in a presence redundancy group when one or more critical services fails on an IM and Presence Service node in the group or a node in the group fails. Clients automatically connect to the other IM and Presence Service node in that group.

• Fallback—Occurs when a fallback command is issued from the CLI or Cisco Unified Communications Manager during either of these conditions:
  • The failed IM and Presence Service node comes back into service and all critical services are running. The failed-over clients in that group reconnect with the recovered node when it becomes available.
  • The backup activated IM and Presence Service node fails due to a critical service failure, and the peer node is in the Failed Over state and supports the automatic recovery fallback.

For example, if you are using presence redundancy groups, Cisco Jabber clients will fail over to a backup IM and Presence Service node if the services or hardware fail on the local IM and Presence Service node. When the failed node comes online again, the clients automatically reconnect to the local IM and Presence Service node if you have configured automatic fallback. If you have not configured automatic fallback, you can manually initiate the fallback when the failed node comes online.

In addition to redundancy and recovery, presence redundancy groups also allow you to configure high availability for your cluster.
High Availability

The IM and Presence Service supports high availability for multiple-node deployments.

After you configure a presence redundancy group, you can enable high availability for the group. A pair of nodes is required for high availability. Each node has an independent database and set of users operating with a shared availability database that is able to support common users.

All IM and Presence Service nodes must belong to a presence redundancy group, which can consist of a single IM and Presence Service node or a pair of IM and Presence Service nodes.

You can configure high availability using two different modes:

- Balanced mode: This mode provides redundant high availability with automatic user load balancing and user failover in the event that one node fails because of component failure or power outage.
- Active/standby mode: The standby node automatically takes over for the active node if the active node fails. It does not provide automatic load balancing.

We recommend that you configure your IM and Presence Service deployments as high availability deployments. Although you are permitted to have both high availability and non-high availability presence redundancy groups configured in a single deployment, this configuration is not recommended.

Presence Redundancy Group Prerequisites

For deployments over the WAN, a minimum of 10 megabits per second of dedicated bandwidth is required for each IM and Presence Service cluster, with no more than an 80-millisecond round-trip latency. Any bandwidth less than this recommendation can adversely impact performance.

Presence Redundancy Group Task Flow

An IM and Presence Service node can be assigned to only one presence redundancy group. For high availability, you must assign two nodes from the same cluster to the presence redundancy group and enable high availability for the group.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Verify Database Replication, on page 397</td>
<td>Ensure that database replication is setup in the IM and Presence Service cluster.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Verify Services, on page 397</td>
<td>Make sure critical services are running on the nodes that you plan to add to a presence redundancy group.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Presence Redundancy Group, on page 398</td>
<td>Provide redundancy and recovery for IM and Presence Service clients and applications.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Heartbeat Interval for Failover, on page 399</td>
<td>Optional. Each node in the presence redundancy group monitors the status, or heartbeat, of its</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------------------------------------------</td>
<td>------------------------------------------------------------------------------------------------------------------------------------------</td>
<td></td>
</tr>
<tr>
<td></td>
<td>peer node. You can configure the intervals by which each node monitors its peer.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Optional. Follow this procedure if you did not enable high availability when you configured the presence redundancy group.</td>
<td></td>
</tr>
<tr>
<td>Enable High Availability, on page 400</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure how you want the Sync Agent to distribute users across various nodes in the IM and Presence Service cluster. This setting affects how your system handles failover and load balancing.</td>
<td></td>
</tr>
<tr>
<td>Configure User Assignment Mode, on page 401</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Verify Database Replication**

Ensure that database replication is setup in the IM and Presence Service cluster before you enable high availability for a presence redundancy group.

**Procedure**

**Step 1**
Start a CLI session using one of the following methods:

- From a remote system, use SSH to connect securely to the Cisco Unified Operating System. In your SSH client, enter your `ssh adminname@hostname` and enter your password.
- From a direct connection to the serial port, enter your credentials at the prompt that displays automatically.

**Step 2**
Execute the `utils dbreplication status` command to check for errors or mismatches in the database tables.

**Step 3**
Execute the `utils dbreplication runtimestate` command to check if the database replication is active on the node.

The output lists all the nodes and if database replication is set up and in a good state, the `replication setup` value for each node is 2.

If a value other than 2 is returned, you must resolve the errors before proceeding.

**What to do next**

Verify Services, on page 397

**Verify Services**

Make sure critical services are running on the nodes that you plan to add to a presence redundancy group. Critical services must be running before you turn on high availability. If critical services are not running on either node, the presence redundancy group will go into a Failed state when you turn on high availability. If critical services are not running on one node, then that node fails over to the other node when you turn on high availability.
Configure a Presence Redundancy Group

Before you begin

Verify Database Replication, on page 397

Procedure

Step 1  From Cisco Unified IM and Presence Serviceability, choose Tools > Control Center - Network Services.
Step 2  From the Server list, choose the appropriate node and click Go.
Step 3  In the IM and Presence Services area, ensure that the following services are started:
   • Cisco Client Profile Agent
   • Cisco Sync Agent
   • Cisco XCP Router
Step 4  From the Related Links drop-down list, select Control Center - Network Services and click Go.
Step 5  In the IM and Presence Services area, ensure that the following services are started:
   • Cisco SIP Proxy
   • Cisco Presence Engine

What to do next
Configure a Presence Redundancy Group, on page 398

Configure a Presence Redundancy Group

Use Cisco Unified Communications Manager to configure redundancy for IM and Presence Service nodes. Each presence redundancy group can contain two IM and Presence Service nodes. Each node can be assigned to only one presence redundancy group. Both nodes in the presence redundancy group must be on the same cluster and have the same IM and Presence Service database publisher node.

Before you begin

• Verify Services, on page 397
• Ensure that the IM and Presence Service nodes you are adding to a presence redundancy group are running the same software version.

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Presence Redundancy Groups.
Step 2  Click Add New.
Step 3  Enter a unique name for the presence redundancy group.
You can enter a maximum of 128 alphanumeric characters, including underscore (_) and dash (-).

**Step 4** Enter a description of the group.

You can enter a maximum of 128 alphanumeric characters including symbols, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), forward slash (/), or angle brackets (<>).

**Step 5** Choose two different IM and Presence Service nodes in the **Presence Server** fields to assign them to the group.

**Step 6** (Optional) Check the **Enable High Availability** check box to enable high availability for the presence redundancy group.

**Step 7** Click **Save**.

---

**What to do next**

Configure Heartbeat Interval for Failover, on page 399

---

**Configure Heartbeat Interval for Failover**

Configure optional service parameters that determine the keep alive settings by which each peer in a presence redundancy group monitors the heartbeat (i.e., the status) of its peer node in order to confirm that the peer is active. A failover can be initiated if the peer node is unresponsive after a configured timer expires.

---

**Note**
Cisco recommends that you use the default values for these service parameters. However, you can also reconfigure the values to suit your needs.

---

**Procedure**

**Step 1** In Cisco Unified CM IM and Presence Administration, choose **System > Service Parameters**.

**Step 2** From the **Server** drop-down, select an IM and Presence node.

**Step 3** From the **Service** drop-down, select **Cisco Server Recovery Manager (Active)**.

**Step 4** Under **General Server Recovery Manager Parameters (Clusterwide)**, configure the clusterwide Keep Alive settings that each node in a Presence Redundancy Group uses to monitor monitor the heartbeat of its peer node. A failover can be initiated if the peer node is unresponsive.

- **Service Port**—This parameter specifies the port that Cisco Server Recovery Manager uses to communicate with its peer. The default is 22001.
- **Admin RPC Port**—This parameter specifies the port that Cisco Server Recovery Manager uses to provide admin rpc requests. The default is 20075.
- **Critical Service Delay**—This parameter specifies the duration in seconds that a critical service can be down before failover is initiated. The default is 90.
- **Enable Automatic Fallback**—This parameter specifies whether to do automatic fallback. In the event of a failover, the IM and Presence Service moves users automatically from the backup node to the primary node thirty minutes after the primary node returns to a healthy state. The default value is False.
- **Initialization Keep Alive (Heartbeat) Timeout**—This parameter specifies the duration in seconds that the heartbeat can be lost with the peer during initialization before failover is initiated. The default is 120.
• **Keep Alive (Heartbeat) Timeout**—This parameter specifies the duration in seconds that the heartbeat can be lost with the peer before failover is initiated. The default is 60.

• **Keep Alive (HeartBeat) Interval**—This parameter specifies the interval in seconds between keep alive (heart beat) messages being sent to the peer. The default is 15.

**Step 5** Configure the following additional parameters, which tell CUPC 8.5 and higher clients how long to wait before attempting to relogin. Unlike the above parameters, these parameters must be configured separately for each cluster node.

• **Client Re-Login Lower Limit**—This parameter specifies the minimum number of seconds which CUPC 8.5 (and higher) should wait before attempting to re-login to this server. The default is 120.

• **Client Re-Login Upper Limit**—This parameter specifies the maximum number of seconds which CUPC 8.5 (and higher) should wait before attempting to re-login to this server. The default is 537.

**Step 6** Click Save.

---

**What to do next**
If you did not enable high availability when you configured the presence redundancy group, Enable High Availability, on page 400 now.

---

**Enable High Availability**

⚠️ **Caution**
Failure to set up replication in the IM and Presence Service cluster and ensure that all critical services are running may result in an immediate failover when high availability is enabled for the presence redundancy group.

---

**Before you begin**

- Configure a Presence Redundancy Group, on page 398

- Ensure that replication is set up in the IM and Presence Service cluster.

- Ensure that all critical services are running.

**Procedure**

1. From Cisco Unified CM Administration, choose System > Presence Redundancy Groups.
2. Specify search criteria and then click Find.
3. Choose the presence redundancy group that you configured.
4. To enable high availability, check the Enable High Availability check box.
5. Click Save.
Configure User Assignment Mode

Use this procedure to configure the way in which the sync agent distributes users to the nodes in the cluster. This setting helps to manage failover and load balancing.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Enterprise Parameters.

Step 2 In the User Management Parameters Area, choose one of the following options for the User Assignment Mode for Presence Server parameter:

- **Balanced**—This mode assigns users equally to each node in each subcluster and attempts to balance the total number of users equally across each node. This is the default option.
- **Active-Standby**—This mode assigns all users to the first node of the subcluster, leaving the secondary server as a backup.
- **None**—This mode results in no assignment of the users to the nodes in the cluster by the sync agent.

Step 3 Click Save.

Redundancy Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Adding Users</td>
<td>You cannot add new users to an IM and Presence Service cluster while one of the cluster nodes is in a failover state.</td>
</tr>
<tr>
<td>Multiple Device Messaging</td>
<td>The Multiple Device Messaging feature causes a delay with server recovery on the IM and Presence Service if failover occurs. If server failover occurs on a system where Multiple Device Messaging is configured, the failover times generally are twice as long as the times specified with the Cisco Server Recovery Manager service parameters.</td>
</tr>
</tbody>
</table>
Interaction

The temporary presence status of a user displays the stale presence status after Failover, Fallback, and user moves. This is because the subscription to temporary presence will be deleted and the user must re-subscribe to temporary presence to see the valid temporary presence status of the user.

For example, if User A is subscribed to user B's temporary presence and a failover occurs on the IM and Presence node where User B is assigned, then user B displays offline to User A even after User B re-logins to the backup node. It is because the subscription to temporary presence of User B is deleted and User A is not aware of the deletion. User A must re-subscribe to temporary presence of User B again.

When User A deletes search of User B from Jabber client, User A needs to wait at least 30 seconds before it tries to search the temporary presence of User B. If not, then User A sees the stale presence of User B. Jabber client must wait for at least 30 seconds between two searches for the same user to get a valid temporary presence status.

## Initiate Manual Failover, Fallback, or Recovery

Use this procedure to initiate manual failover, fallback, or recovery of IM and Presence Service nodes within a presence redundancy group.

- Manual failover—When you initiate a manual failover, the Cisco Server Recovery Manager stops the critical services on the failed node. All users from the failed node are disconnected and must re-login to the backup node. Critical services will not be restarted unless we invoke manual fallback.

- Manual fallback—When you initiate a manual fallback, the Cisco Server Recovery Manager restarts critical services on the primary node and disconnects all users that had been failed over. Those users must then re-login to their assigned node.

- Manual recovery—A manual recovery is necessary when both nodes in the presence redundancy group are in the failed state. In this case, the IM and Presence Service restarts the Cisco Server Recovery Manager service on both nodes in the presence redundancy group.

## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose System &gt; Presence Redundancy Groups.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select the Presence Redundancy Group with the applicable nodes.</td>
</tr>
</tbody>
</table>
| Step 3 | Do one of the following. Note that the available button depends on the current state of the node:  
  - Click Failover to initiate failover of an active node.  
  - Click Fallback to initiate fallback of a failed over node.  
  - Click Recover if both nodes are failed over and you want to recover them. |
You can also initiate these actions from Cisco Unified Communications Manager or IM and Presence Service using the CLI. See the Command Line Interface Guide for Cisco Unified Communications Solutions for details.

Note
You cannot add end users to an IM and Presence Service cluster while one of the nodes is in a failover state.

Node State Definitions

Table 64: Presence Redundancy Group Node State Definitions

<table>
<thead>
<tr>
<th>State</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Initializing</td>
<td>This is the initial (transition) state when the Cisco Server Recovery Manager service starts; it is a temporary state.</td>
</tr>
<tr>
<td>Idle</td>
<td>IM and Presence Service is in Idle state when failover occurs and services are stopped. In Idle state, the IM and Presence Service node does not provide any availability or Instant Messaging services. In Idle state, you can manually initiate a fallback to this node using the Cisco Unified CM Administration user interface.</td>
</tr>
<tr>
<td>Normal</td>
<td>This is a stable state. The IM and Presence Service node is operating normally. In this state, you can manually initiate a failover to this node using the Cisco Unified CM Administration user interface.</td>
</tr>
<tr>
<td>Running in Backup Mode</td>
<td>This is a stable state. The IM and Presence Service node is acting as the backup for its peer node. Users have moved to this (backup) node.</td>
</tr>
<tr>
<td>Taking Over</td>
<td>This is a transition state. The IM and Presence Service node is taking over for its peer node.</td>
</tr>
<tr>
<td>Failing Over</td>
<td>This is a transition state. The IM and Presence Service node is being taken over by its peer node.</td>
</tr>
<tr>
<td>Failed Over</td>
<td>This is a steady state. The IM and Presence Service node has failed over, but no critical services are down. In this state, you can manually initiate a fallback to this node using the Cisco Unified CM Administration user interface.</td>
</tr>
<tr>
<td>Failed Over with Critical Services Not Running</td>
<td>This is a steady state. Some of the critical services on the IM and Presence Service node have either stopped or failed.</td>
</tr>
<tr>
<td>Falling Back</td>
<td>This is a transition state. The system is falling back to this IM and Presence Service node from the node that is running in backup mode.</td>
</tr>
<tr>
<td>Taking Back</td>
<td>This is a transition state. The failed IM and Presence Service node is taking back over from its peer.</td>
</tr>
<tr>
<td>Running in Failed Mode</td>
<td>An error occurs during the transition states or Running in Backup Mode state.</td>
</tr>
</tbody>
</table>
### Node States, Causes, and Recommended Actions

You can view the status of nodes in a presence redundancy group on the **Presence Redundancy Group Configuration** window when you choose a group using the **Cisco Unified CM Administration** user interface.

**Table 65: Presence Redundancy Group Node High-Availability States, Causes, and Recommended Actions**

<table>
<thead>
<tr>
<th>Node 1 State</th>
<th>Node 2 State</th>
<th>Cause/Recommended Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal</td>
<td>Normal</td>
<td>Normal</td>
</tr>
<tr>
<td>Failing Over</td>
<td>On Admin Request</td>
<td>On Admin Request</td>
</tr>
<tr>
<td>Idle</td>
<td>On Admin Request</td>
<td>Running in Backup Mode</td>
</tr>
<tr>
<td>Taking Back</td>
<td>On Admin Request</td>
<td>Falling Back</td>
</tr>
<tr>
<td>Idle</td>
<td>Initialization</td>
<td>Running in Backup Mode</td>
</tr>
<tr>
<td>Idle</td>
<td>Initialization</td>
<td>Running in Backup Mode</td>
</tr>
<tr>
<td>Idle</td>
<td>On Admin Request</td>
<td>Running in Backup Mode</td>
</tr>
<tr>
<td>Failing Over</td>
<td>On Admin Request</td>
<td>Taking Over</td>
</tr>
<tr>
<td>Taking Back</td>
<td>Initialization</td>
<td>Falling Back</td>
</tr>
<tr>
<td>Node 1</td>
<td>Node 2</td>
<td>Cause/Recommended Actions</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-------------------------------</td>
<td>-------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Taking Back</strong></td>
<td><strong>Automatic Fallback</strong></td>
<td>Automatic Fallback has been initiated from node 2 to node 1 and is currently in progress.</td>
</tr>
<tr>
<td><strong>Failed Over</strong></td>
<td><strong>Initialization or Critical Services Down</strong></td>
<td><strong>Running in Backup Mode</strong></td>
</tr>
<tr>
<td><strong>Failed Over with Critical Services not Running</strong></td>
<td><strong>Critical Service Down</strong></td>
<td><strong>Running in Backup Mode</strong></td>
</tr>
<tr>
<td><strong>Failed Over with Critical Services not Running</strong></td>
<td><strong>Database Failure</strong></td>
<td><strong>Running in Backup Mode</strong></td>
</tr>
</tbody>
</table>

**Critical Service Down**

A database service is down on node 1. IM and Presence Service performs an automatic failover to node 2.

**Recommended Actions:**

1. Reboot node 1.

2. When all the critical services are up and running after the reboot, perform a manual fallback to restore the nodes in the presence redundancy group to the Normal state.

**Critical Service Down**

<table>
<thead>
<tr>
<th>Critical Service Down</th>
<th>Running in Backup Mode</th>
<th>Critical Service Down</th>
</tr>
</thead>
</table>

A critical service is down on node 1. IM and Presence Service performs an automatic failover to node 2.

**Recommended Actions:**

1. Check node 1 for any critical services that are down and try to manually start those services.

2. If the critical services on node 1 do not start, then reboot node 1.

3. When all the critical services are up and running after the reboot, perform a manual fallback to restore the nodes in the presence redundancy group to the Normal state.

**Failed Over with Critical Services not Running**

- Critical services come back up due to a reboot of node 1.
- The administrator starts critical services on node 1 while node 1 is in Failed Over with Critical Services Not Running state.

When node 1 transitions to Failed Over state the node is ready for the administrator to perform a manual fallback to restore the nodes in the presence redundancy group to Normal state.

**Initialization or Critical Services Down**

**Critical Service Down**

<table>
<thead>
<tr>
<th>Initialization or Critical Services Down</th>
<th>Running in Backup Mode</th>
<th>Critical Service Down</th>
</tr>
</thead>
</table>

**Failed Over**

Node 1 transitions to Failed Over state when either of the following conditions occur:

- Critical services come back up due to a reboot of node 1.
- The administrator starts critical services on node 1 while node 1 is in Failed Over with Critical Services Not Running state.

<table>
<thead>
<tr>
<th>Automatic Fallback</th>
<th>Automatic Fallback</th>
<th>Automatic Fallback</th>
</tr>
</thead>
</table>

**Automatic Fallback**

Automatic Fallback has been initiated from node 2 to node 1 and is currently in progress.
### Node States, Causes, and Recommended Actions

<table>
<thead>
<tr>
<th>Node 1 State</th>
<th>Reason</th>
<th>Node 2 State</th>
<th>Reason</th>
<th>Cause/Recommended Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Running in Failed Mode</td>
<td>Start of Critical Services Failed</td>
<td>Running in Failed Mode</td>
<td>Start of Critical Services Failed</td>
<td>Critical services fail to start while a node in the presence redundancy group is taking back from the other node.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td><strong>Recommended Actions.</strong> On the node that is taking back, perform the following actions:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1. Check the node for critical services that are down. To manually start these services, click <strong>Recovery</strong> in the <strong>Presence Redundancy Group Configuration</strong> window.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2. If the critical services do not start, reboot the node.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3. When all the critical services are up and running after the reboot, perform a manual fallback to restore the nodes in the presence redundancy group to the Normal state.</td>
</tr>
<tr>
<td>Running in Failed Mode</td>
<td>Critical Service Down</td>
<td>Running in Failed Mode</td>
<td>Critical Service Down</td>
<td>Critical services go down on the backup node. Both nodes enter the failed state.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td><strong>Recommended Actions:</strong></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1. Check the backup node for critical services that are down. To start these services manually, click <strong>Recovery</strong> in the <strong>Presence Redundancy Group Configuration</strong> window.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2. If the critical services do not start, reboot the node.</td>
</tr>
<tr>
<td>Node 1</td>
<td>Node 2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>--------</td>
<td>--------</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>State</td>
<td>Reason</td>
<td>State</td>
<td>Reason</td>
<td>Cause/Recommended Actions</td>
</tr>
<tr>
<td>Node 1 is down due to loss of network connectivity or the SRM service is not running.</td>
<td>Running in Backup Mode</td>
<td>Peer Down</td>
<td>Node 2 has lost the heartbeat from node 1. IM and Presence Service performs an automatic failover to node 2.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Peer Reboot</td>
<td>IM and Presence Service performs an automatic failover to node 2 due to the following possible conditions on node 1:</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>• hardware failure</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>• power down</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>• restart</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>• shutdown</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Recommended Actions:**

1. Repair and power up node 1.
2. When the node is up and all critical services are running, perform a manual fallback to restore the nodes in the presence redundancy group to the Normal state.
### Node States, Causes, and Recommended Actions

<table>
<thead>
<tr>
<th>Node 1</th>
<th>Node 2</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>State</strong></td>
<td><strong>Reason</strong></td>
</tr>
<tr>
<td>Failed Over with Critical Services not Running OR Failed Over</td>
<td>Initialization</td>
</tr>
<tr>
<td>Running in Failed Mode</td>
<td>Cisco Server Recovery Manager Take Over Users Failed</td>
</tr>
<tr>
<td>Running in Failed Mode</td>
<td>Cisco Server Recovery Manager Take Back Users Failed</td>
</tr>
<tr>
<td>Running in Failed Mode</td>
<td>Unknown</td>
</tr>
<tr>
<td>Backup Activated</td>
<td>Auto Recover Database Failure</td>
</tr>
<tr>
<td>Backup Activated</td>
<td>Auto Recover Database Failure</td>
</tr>
<tr>
<td>Node 1</td>
<td>Node 2</td>
</tr>
<tr>
<td>-------</td>
<td>-------</td>
</tr>
<tr>
<td><strong>State</strong></td>
<td><strong>Reason</strong></td>
</tr>
<tr>
<td>Unknown</td>
<td>Unknown</td>
</tr>
</tbody>
</table>
Configure Cisco Unity Connection for Voicemail and Messaging

Cisco Unity Connection

As you start configuring your voicemail and messaging system, be aware of the options that you have for adding users, enabling features, and integrating Cisco Unified Communications Manager with Cisco Unity Connection.

When integrated with Cisco Unified Communications Manager, Cisco Unity Connection (the voicemail and messaging system) provides voice-messaging features for users that you configure manually, through AXL services, or through LDAP integration. After receiving voice messages in their mailboxes, users receive message-waiting lights on their phones. Users can retrieve, listen to, reply to, forward, and delete their messages by accessing the voice-messaging system with an internal or external call.

Your system supports both directly connected and gateway-based messaging systems. Directly connected voice-messaging systems communicate with Cisco Unified Communications Manager by using a packet protocol. A gateway-based voice-messaging system connects to Cisco Unified Communications Manager through analog or digital trunks that then connect to Cisco gateways.

When you integrate Unified Communications Manager and Cisco Unity Connection, you can configure the following features for your users:
  - Call forward to personal greeting
  - Call forward to busy greeting
  - Caller ID
  - Easy message access (a user can retrieve messages without entering an ID; Cisco Unity Connection identifies a user based on the extension from which the call originated; a password may be required)
  - Identified user messaging (Cisco Unity Connection automatically identifies a user who leaves a message during a forwarded internal call, based on the extension from which the call originated)
  - Message waiting indication (MWI)
• The configuration of a secure SIP trunk integration between a Cisco Unified Communications Manager and Cisco Unity Connection server requires that the Cisco Unified Communications Manager cluster is configured in mixed mode.

Cisco Unified Communications Manager interacts with Cisco Unity Connection through one of the following interfaces:

• SIP Trunk—You can integrate Cisco Unity Connection and Unified Communications Manager by using SIP. Instead of multiple SCCP ports involved with traditional integrations, SIP uses a single trunk per Unity Connection server. The SIP integration eliminates the requirement to configure directory numbers for Voicemail Ports and message-waiting indicators (MWIs).

• SCCP Protocol—You configure the interface to directly connected voice-messaging systems by creating voicemail ports. These establish a link between Unified Communications Manager and Cisco Unity Connection.

To handle multiple, simultaneous calls to a voice-messaging system, you create multiple voicemail ports and place the ports in a line group and the line group in a route/hunt list.

Cisco Unified Communications Manager generates SCCP messages, which are translated by Cisco Unity Connection. The voicemail system sends message-waiting indications (MWIs) by calling a message-waiting on and off number.

When you configure security for voicemail ports and Cisco Unity SCCP devices, a TLS connection (handshake) opens for authenticated devices after each device accepts the certificate of the other device; likewise, the system sends SRTP streams between devices; that is, if you configure the devices for encryption.

When the device security mode is set to authenticated or encrypted, the Cisco Unity TSP connects to Cisco Unified Communications Manager through the Unified Communications Manager TLS port. When the security mode is nonsecure, the Cisco Unity TSP connects to Cisco Communications Manager through the Unified Communications Manager SCCP port.


Cisco Unity Connection for Voicemail and Messaging Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Cisco Unity Connection for Voicemail and messaging.</td>
<td>To configure Cisco Unity Connection, see the Cisco Unified Communications Manager SCCP Integration Guide for Cisco Unity Connection or the Cisco Unified Communications Manager SIP Trunk Integration Guide for Cisco Unity Connection</td>
</tr>
</tbody>
</table>
Enable PIN Synchronization

Use this procedure to enable PIN synchronization so that the end users can log in to Extension Mobility, Conference Now, Mobile Connect, and the Cisco Unity Connection Voicemail using the same PIN.

The pin synchronization between Cisco Unity Connection and Cisco Unified Communications Manager is successful, only when Cisco Unified Communications Manager publisher database server is running and completes its database replication. Following error message is displayed when the pin synchronization fails on Cisco Unity Connection: Failed to update PIN on CUCM. Reason: Error getting the pin.

If the PIN Synchronization is enabled and the end user changes the pin, then pin is updated in Cisco Unified Communications Manager. This happens only when the pin update is successful in at least one of the configured Unity Connection Application servers.

For PIN Synchronization to take effect, administrators must force the users to change their PIN after successfully enabling the feature.

Before you begin

This procedure assumes that you already have your application server connection to Cisco Unity Connection setup. If not, for more information on how to add a new application server, see the Related Topics section.

To enable PIN Synchronization feature, you need to first upload a valid certificate for the Cisco Unity Server connection from the Cisco Unified OS Administration page to the Cisco Unified Communications Manager tomcat-trust. For more information on how to upload the certificate, see the “Manage Security Certificates” chapter in the Administration Guide for Cisco Unified Communications Manager at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

The user ID in the Cisco Unity Connection Server must match the user ID in Cisco Unified Communications Manager.

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Application Servers.
**Step 2** Select the application server that you set up for Cisco Unity Connection.

**Step 3** Check the **Enable End User PIN Synchronization** check box.

**Step 4** Click **Save**.

**Related Topics**
- [Configure Application Servers](#), on page 388
Configure Cisco Unified Contact Center Enterprise

You can use Cisco Unified Contact Center Enterprise (Unified CCE) in your system to integrate intelligent call routing, network-to-desktop computer telephony integration (CTI), and multichannel contact management to contact center agents over an IP network. Unified CCE combines software IP automatic call distribution (ACD) with Cisco Unified Communications so that you can rapidly deploy an advanced, distributed contact center.

CHAPTER 53

Configure Cisco Unified Contact Center Express

Cisco Unified Contact Center Express

Cisco Unified Contact Center Express (Unified CCX) provides your system with the features of a large contact center packaged into a single- or dual-server deployment. Unified CCX scales up to 400 concurrent agents, 42 supervisors, 150 agent groups, and 150 skill groups. It includes email, chat, outbound calling, inbound calling, workforce optimization, and reporting.

Unified CCX works with Unified Communication Manager, which manages all contact center calls on behalf of Unified CCX. When a call is placed to your help desk, your call system recognizes that the number is destined for the Unified CCX application server. With this configuration, Unified CCX receives the incoming call and handles the request based on the extension number that was dialed. The script plays prompts, collects digits and, if necessary, uses the information from the caller to select an appropriate agent. If an assigned agent is not available, the call is put into an appropriate queue and a recorded message or music is streamed to the caller. As soon as an agent is available, Unified CCX instructs Unified Communications Manager to ring the agent’s phone.

When the agent picks up, relative call context is provided in the agent’s desktop application. This step ensures that agents have the proper information in front of them to support the customer.

Configure CTI Applications

CTI Applications Overview

You can use Computer Telephony Integration (CTI) to take advantage of computer-processing functions while making, receiving, and managing telephone calls. CTI applications allow you to perform such tasks as retrieving customer information from a database using a caller ID, or to work with the information gathered by an Interactive Voice Response (IVR) system to route a customer’s call, along with their information, to the appropriate customer service representative.

Applications that want to terminate media for calls at route points must specify the media and port for the call on a per-call basis. CTI applications can terminate media on CTI ports and CTI route points using either static or dynamic IP addresses and port numbers.

This chapter describes how to configure Cisco Unified Communications Manager to work with CTI applications. For information about how to configure specific applications, see the Feature Configuration Guide for Cisco Unified Communications Manager.

Some of the Cisco CTI applications available are:

- Cisco IP Communicator: A desktop application which turns your computer into a full-feature telephone with the added advantages of call tracking, desktop collaboration, and one-click dialing from online directories.

- Cisco Unified Communications Manager Auto-Attendant: Works with Unified Communications Manager to receive calls on specific telephone extensions and to allow the caller to choose an appropriate extension.

- Cisco Web Dialer: Allows Cisco IP Phone users to make calls from web and desktop applications.

- Cisco Unified Communications Manager Assistant: Enables managers and their assistants to work together more effectively. The feature comprises a call-routing service, enhancements to phone capabilities for the manager and the assistant, and assistant console interfaces that are primarily used by the assistant.

Note

To determine which Unified Communications Manager CTI applications support SIP IP phones, see the application-specific documentation.
CTI Route Points Overview

A CTI route point virtual device can receive multiple, simultaneous calls for application-controlled redirection. You can configure one or more lines on a CTI route point that users can call to access the application. Applications can answer calls at a route point and can also redirect calls to a CTI port or IP phone. When a CTI application requests to redirect a call by using the Redirect API, Cisco Unified Communications Manager uses the configuration for the line/device calling search space for the redirected party.

With CTI route points you can:

- Answer a call
- Make and receive multiple active calls
- Redirect a call
- Hold a call
- Unhold a call
- Drop a call

CTI Redundancy on Cisco Unified Communications Manager

When a Unified Communications Manager node in a cluster fails, the CTIManager recovers the affected CTI ports and route points by reopening these devices on another Unified Communications Manager node. If an application has a phone device open, the CTIManager also reopens the phone when the phone fails over to a different Unified Communications Manager. If the Cisco IP Phone does not fail over to a different Unified Communications Manager, the CTIManager cannot open the phone or a line on the phone. The CTIManager uses the Unified Communications Manager group that is assigned to the device pool to determine which Unified Communications Manager to use to recover the CTI devices and phones that the applications opened.

CTI Redundancy on CTIManager

When a CTIManager fails, the applications that are connected to the CTIManager can recover the affected resources by reopening these devices on another CTIManager. An application determines which CTIManager to use on the basis of CTIManagers that you defined as primary and backup when you set up the application (if supported by the application). When the application connects to the new CTIManager, it can reopen the devices and lines that previously opened. An application can reopen a Cisco IP Phone before the phone rehomes to the new Unified Communications Manager; however, it cannot control the phone until the rehoming completes.

Note

The applications do not rehome to the primary CTIManager when it comes back in service. Applications fail back to the primary CTIManager if you restart the application or if the backup CTIManager fails.

CTI Redundancy for Application Failure

When an application (TAPI/JTAPI or an application that directly connects to the CTIManager) fails, the CTIManager closes the application and redirects unterminated calls at CTI ports and route points to the
configured call forward on failure (CFOF) number. The CTIManager also routes subsequent calls into those CTI ports and route points to the configured Call Forward No Answer (CFNA) number until the application recovers and reregisters those devices.

### CTI Applications Prerequisites

You must have device pools configured before you can configure Cisco Unified Communications Manager for CTI Applications.

Add and configure IP phones for each CTI application. For further information on adding and configuring IP Phones see, Cisco Unified IP Phones.

Configure the end users and application users that will use CTI applications.

Computer Telephony Integration (CTI) provides IP address information through the JTAPI and TAPI interfaces, which can support IPv4 and IPv6 addresses. If you want to support IPv6 addresses, make sure that your applications are using a JTAPI / TAPI client interface version that supports IPv6.

### Configure CTI Applications Task Flow

To configure Cisco Unified Communications Manager for CTI applications follow these tasks.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Activate the CTIManager Service, on page 422</td>
<td>Activate the CTIManager service on the appropriate servers, if not already activated.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure CTIManager and Cisco Unified Communications Manager Service Parameters, on page 422</td>
<td>Configure CTIManager advanced clusterwide service parameters that are used in conjunction with the CTI Super Provider capability.</td>
</tr>
</tbody>
</table>
| **Step 3** | To configure CTI Route Points perform the following procedure:  
  - Configure CTI Route Points, on page 423  
  - Configure New Call Accept Timer, on page 423  
  - Configure Simultaneous Active Calls, on page 424  
  - Synchronize CTI Route Point, on page 424 | Configure one or more CTI route point virtual devices which can receive multiple, simultaneous calls for application-controlled redirection. |
| **Step 4** | Configure CTI Device Directory Number, on page 425 | Configure the directory number for the CTI device. |
| **Step 5** | Associate Devices with Groups, on page 425 | Associate all devices that the application will use for application users and end users with the appropriate Cisco Unified Communications Manager group (via the device pool). |
Activate the CTIManager Service

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 6</td>
<td>Add End Users and Application Users, on page 425</td>
<td>Allow a CTI application to control any CTI-controllable devices that are configured in the Cisco Unified Communications Manager system by adding the end users and application users to the Standard CTI Enabled user group.</td>
</tr>
<tr>
<td>Step 7</td>
<td>(Optional) Configure CTI Redundancy for Application Failure, on page 426</td>
<td>To define the interval at which CTIManager expects to receive a message from an application within two consecutive intervals.</td>
</tr>
</tbody>
</table>

### Configure CTIManager and Cisco Unified Communications Manager Service Parameters

Configure CTIManager advanced clusterwide service parameters that are used in conjunction with the CTI Super Provider capability.

**Note**
If the configured limits are exceeded, CTI generates alarms, but the applications continue to operate with the extra devices.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose System &gt; Service Parameters.</td>
<td>Configure the service parameters.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Choose the Server</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose Cisco CTIManager (Active) from the Service</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>On the Service Parameter Configuration window, click Advanced.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>In the Maximum Devices Per Provider field, enter the maximum number of devices that a single CTI application can open. The default is 2000 devices.</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>In the Maximum Devices Per Node field, enter the maximum number of devices that all CTI applications can open on any CTIManager node in the Unified Communications Manager system. The default is 800 devices.</td>
<td></td>
</tr>
</tbody>
</table>
Configure CTI Route Points Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure CTI Route Points, on page 423</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure New Call Accept Timer, on page 423</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Simultaneous Active Calls, on page 424</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Optional: Synchronize CTI Route Point, on page 424</td>
</tr>
</tbody>
</table>

**Configure CTI Route Points**

Add a new, or modify an existing CTI route point.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, click **Device > CTI Route Point**.
- **Step 2** Perform one of the following tasks:
  - Click **Add New**, to add a new gateway.
  - Click **Find** and select a CTI route point from the resulting list to modify the settings for an existing CTI route point, enter search criteria.
- **Step 3** Configure the fields in the **CTI Route Point Configuration** window. For more information on the fields and their configuration options, see the system Online Help.
- **Step 4** Click **Save**.

**Configure New Call Accept Timer**

Configure the New Call Accept Timer so that when a call arrives at a route point, the application will handle (accept, answer, redirect) it within the time specified.
Procedure

Step 1  From Cisco Unified CM Administration, choose System > Service Parameters.
Step 2  Choose the node from the Server drop-down list.
Step 3  Choose Cisco CallManager (Active) from the Service drop-down list.
Step 4  In the CTI New Call Accept Timer field, specify the time that you want to allow for a call to be answered. The default value is 4.
Step 5  Click Save.

Configure Simultaneous Active Calls

Configure the number of simultaneous active calls on the route point.

Note

If you are planning to use a TAPI application to control CTI port devices by using the Cisco CallManager Telephony Service Provider (TSP), you may only configure one line per CTI port device.

Procedure

Step 1  From Cisco Unified CM Administration, click Call Routing > Directory Number.
Step 2  On the Directory Number Configuration window, click Add New.
Step 3  Fill in the required fields.
Step 4  Click Save.

Synchronize CTI Route Point

Synchronize a CTI route point with the most recent configuration changes, which applies any outstanding configuration settings in the least intrusive manner possible. (For example, a reset/restart may not be required on some affected devices.)

Procedure

Step 1  From Cisco Unified CM Administration, click Device > CTI Route Point.
Step 2  On the Find and List CTI Route Points window, click Find to display the list of CTI route points.
Step 3  Check the check boxes next to the CTI route points that you want to synchronize. To choose all CTI route points in the window, check the check box in the matching records title bar.
Step 4  Click Apply Config to Selected.
Step 5  Click OK.
Configure CTI Device Directory Number

Configure the directory number for the CTI device.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Call Routing &gt; Directory Number.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>On the Find and List Directory Numbers window, click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>On the Directory Number Configuration window, and enter the required fields.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Associate Devices with Groups

Associate all devices that the application will use for application users and end users with the appropriate Cisco Unified Communications Manager group (via the device pool).

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, click User Management &gt; Application User.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>On the Find and List Application Users window, click Add New. This brings you to the Application User Configuration window.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Device Information pane, associate your devices by moving them from the Available Devices list to the Controlled Devices list.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 5</td>
<td>To Associate Devices for end users, click User Management &gt; End User.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Repeat steps 2 - 4.</td>
</tr>
</tbody>
</table>

Add End Users and Application Users

Allow a CTI application to control any CTI-controllable devices that are configured in the Cisco Unified Communications Manager system by adding the end users and application users to the Standard CTI Enabled user group.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, click User Management &gt; User Settings &gt; Access Control Group.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>On the Find and List Access Control Groups window, click Find to display the current list of access control groups.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Standard CTI Enabled, this brings you to the Access Control Group Configuration window for this group. Ensure all CTI users are in the Standard CTI Enabled user group. See Access Control Group Configuration Options, for a full list of available groups and their capabilities.</td>
</tr>
</tbody>
</table>
Step 4 If you want to add end users, click Add End Users to Group or, if you want to add application users, click Add App Users to Group.

Step 5 Click Find, to display the list of current users.

Step 6 Check the users you want to assign to the Standard CTI Enabled user group.

Step 7 Click Add Selected.

Access Control Group Configuration Options

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standard CTI Allow Call Monitoring</td>
<td>This user group allows an application to monitor calls.</td>
</tr>
<tr>
<td>Standard CTI Allow Call Park Monitoring</td>
<td>This user group allows an application to receive a notification when calls are parked/unparked to all Call Park directory numbers.</td>
</tr>
<tr>
<td>Standard CTI Allow Call Recording</td>
<td>This user group allows an application to record calls.</td>
</tr>
<tr>
<td>Standard CTI Allow Calling Number Modification</td>
<td>This user group allows an application to modify the calling party number in supported CTI applications.</td>
</tr>
<tr>
<td>Standard CTI Allow Control of All Devices</td>
<td>This user group allows an application to control or monitor any CTI-controllable device in the system.</td>
</tr>
<tr>
<td>Standard CTI Allow Reception of SRTP Key Material</td>
<td>This user group allows an application to receive information that is necessary to decrypt encrypted media streams. This group typically gets used for recording and monitoring purposes.</td>
</tr>
<tr>
<td>Standard CTI Enabled</td>
<td>This user group, which is required for all CTI applications, allows an application to connect to Cisco Unified Communications Manager and to access CTI functionality.</td>
</tr>
<tr>
<td>Standard CTI Secure Connection</td>
<td>Inclusion into this group requires that the application has a secure (TLS) CTI connection to Cisco Unified Communications Manager and that the Cisco Unified Communications Manager cluster has security enabled.</td>
</tr>
</tbody>
</table>

Configure CTI Redundancy for Application Failure

To define the interval at which CTI Manager expects to receive a message from an application within two consecutive intervals.
### Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>System &gt; Service Parameters</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Choose the node from the <strong>Server</strong> drop-down list.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose <strong>Cisco CTIManager (Active)</strong> from the <strong>Service</strong> drop-down list.</td>
</tr>
<tr>
<td>Step 4</td>
<td>On the <strong>Service Parameter Configuration</strong> window, click <strong>Advanced</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the <strong>Application Heartbeat Minimum Interval</strong> field, enter the time for the minimum interval. The default is 5.</td>
</tr>
<tr>
<td>Step 6</td>
<td>In the <strong>Application Heartbeat Maximum Interval</strong> field, enter the time for the maximum interval. The default is 3600.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>
Configure CTI Redundancy for Application Failure
Configure Cisco TelePresence

Cisco TelePresence

Cisco TelePresence Conductor

The Cisco TelePresence Conductor simplifies multiparty video communications. The conductor lies within a video communications network, working in conjunction with one or more conference bridges and one or more call control devices (either Cisco TelePresence Video Communication Servers (VCSs) or Unified Communications Managers). It allows the video network to be configured so you can spontaneous or rendezvous conferences can be easily provisioned, initiated, accessed, and managed.

For ad hoc conferences, a SIP trunk is used from Unified Communications Manager to TelePresence Conductor. Set up the relevant TelePresence Conductor Location’s ad hoc IP address as the destination of a SIP trunk on Unified Communications Manager. Ad hoc calls for that location can then be routed down that SIP trunk.

For rendezvous conferences a separate SIP trunk is used from Unified Communications Manager to TelePresence Conductor. Set up the relevant TelePresence Conductor Location’s rendezvous IP address as the destination of a SIP trunk on Unified Communications Manager. Rendezvous calls for that location can then be routed down that SIP trunk.


Cisco TelePresence Conference Bridges

The Cisco TelePresence Server is a scalable videoconferencing bridge that works with Cisco Unified Communications Manager to bring multiparty video to your unified communications deployments. It offers flexible video, audio, and content-sharing capabilities for multiparty videoconferencing. You can easily create, launch, and join meetings using standards-based video endpoints, mobile devices, Cisco WebEx clients, and third-party video endpoints.

The Cisco TelePresence Multipoint Control Unit (MCU) is a high definition multipoint video conferencing bridge. It delivers up to 1080p at 30 frames per second, full continuous presence for all conferences, full transcoding, and is ideal if you want to configure mixed high definition endpoint environments. The Cisco
TelePresence MCU supports SIP as the signaling call control protocol. It has a built-in Web Server that allows for complete configuration, control, and monitoring of the system and conferences.

The Cisco TelePresence Server is primarily controlled by Cisco TelePresence Conductor. For detailed tasks about how to configure these conference bridges within your system, see the deployment guides at http://www.cisco.com/c/en/us/support/conferencing/telepresence-conductor/products-installation-and-configuration-guides-list.html.

**Cisco TelePresence Video Communication Server**

The Cisco TelePresence Video Communication Server (VCS) simplifies session management and control of telepresence conferences. The VCS delivers secure communications, simplified large-scale provisioning, and network administration in conjunction with Cisco TelePresence Management Suite (Cisco TMS). The VCS interworks with Cisco Unified Communications Manager (Unified Communications Manager), bringing rich telepresence services to your system.

Configure Cisco Jabber

Cisco Jabber is a suite of Unified Communications applications that allow seamless interaction with your contacts from anywhere. Cisco Jabber offers IM, presence, audio and video calling, voicemail, and conferencing.

The applications in the Cisco Jabber family of products are:

- Cisco Jabber for Windows
- Cisco Jabber for Mac
- Cisco Jabber for iPhone and iPad
- Cisco Jabber for Android
- Cisco Jabber Softphone for VDI


Cisco Jabber Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Graceful registration</td>
<td>Graceful registration covers dual registration attempts from two Cisco Jabber clients with the same device name (for example, Jabber running on both an office laptop and a home office laptop). The feature de-registers the initial registration automatically so that the second registration can proceed. The de-registered Jabber client does not re-register. Graceful registration is supported automatically for Cisco Jabber, except when Jabber is deployed in a Mobile and Remote Access (MRA) deployment. In MRA deployments, the de-registered Jabber client attempts to re-register. For MRA deployments, if you have Cisco Jabber running on two devices with the same device name, make sure to log Jabber out of one device before you use the other.</td>
</tr>
</tbody>
</table>
PART VIII

Configure Media Resources

- Media Resources Overview, on page 435
- Define Media Resources, on page 437
- Configure Trusted Relay Points, on page 443
- Configure Annunciator, on page 451
- Configure Interactive Voice Response, on page 459
- Configure Video On Hold Server, on page 465
- Configure Announcements, on page 469
- Configure Conference Bridges, on page 473
- Configure Flexible DSCP Marking and Video Promotion, on page 481
- Configure Transcoders and Media Termination Points, on page 489
Media Resources Overview

- About Media Resources, on page 435
- Media Resources Configuration Task Flow, on page 435

About Media Resources

Cisco Unified Communications Manager functionality requires the use of media resources. Cisco Unified Communications Manager includes media resources such as:

- Annunciators
- Interactive Voice Response (IVR)
- Media Termination Points (MTP)
- Transcoders
- Trusted Relay Points
- Conference Bridges
- Music On Hold/Video on Hold

You can make media resources available to calls by assigning them to a media resource group list, and then assigning that list to a device pool, or to an individual device. The default setting for individual devices is to use the media resources that are assigned to the device pool that the device is using.

For information on configuring Music On Hold, refer to the Feature Configuration Guide for Cisco Unified Communications Manager.

Media Resources Configuration Task Flow

Complete the following task flows to configure media resources for your system.
## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Media Resource Group Task Flow, on page 438</td>
<td>Use the procedures in this chapter to define logical groupings of media servers.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Trusted Relay Points Task Flow, on page 444</td>
<td>Insert trusted relay points into a media stream to act as a control point for that stream. This device provides further processing on that stream or as a method to ensure that the stream follows a specific path.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Annunciator Configuration Task Flow, on page 453</td>
<td>Configure the annunciator to enable Unified Communications Manager to play prerecorded announcements (.wav files) and to send tones to devices such as Cisco IP Phones and gateways that are configured for Cisco Multilevel Precedence and Preemption.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Interactive Voice Response Configuration Task Flow, on page 461</td>
<td>Use the Interactive Voice Response (IVR) device to play prerecorded feature announcements (.wav files) to devices such as Cisco IP Phones and Gateways. These announcements play on devices that use features which require IVR announcements, like Conference Now.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Video on Hold Configuration Task Flow, on page 466</td>
<td>Configure video on hold in video contact centers where customers calling into the video contact center are able to watch a specific video after initial consultation with the agent at the contact center.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Announcements Configuration Task Flow, on page 470</td>
<td>Use the procedures in this chapter to use pre-defined announcements or upload custom announcements.</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Conference Bridge Configuration Task Flow, on page 479</td>
<td>Configure software and hardware applications that allow ad hoc and meet-me voice conferencing, as well as video conferencing.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>DSCP Settings Configuration Task Flow, on page 483</td>
<td>Use flexible DSCP marking and video promotion to configure a policy that specifies which applications receive the most favorable call admission control (CAC) and quality of service (QoS) treatment.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>Transcoders and MTPs Configuration Task Flow, on page 494</td>
<td>Configure transcoders to convert an input stream from one codec into an output stream that uses a different codec.</td>
</tr>
</tbody>
</table>
CHAPTER 58

Define Media Resources

- Media Resource Group Overview, on page 437
- Media Resource Group List, on page 438
- Media Resource Group Prerequisites, on page 438
- Media Resource Group Task Flow, on page 438
- Media Resource Group Interactions and Restrictions, on page 441

Media Resource Group Overview

Media resource groups define logical groupings of media servers. You can associate a media resource group with a geographical location or a site, as desired. You can also form media resource groups to control the usage of servers or the type of service (unicast or multicast) that is required.

The system has a two-tiered approach to managing media resources:

- Media resource groups—A logical grouping of media servers.

- Media resource group lists—A prioritized list of media resource groups. An application selects the required media resource, such as a music on hold server from the available media resources according to the priority order that you define in a media resource group list. Media resource group lists, which are associated with devices, provide media resource group redundancy.

You can group devices of the following types into a media resource group:

- Conference Bridge (CFB)
- Media Termination Point (MTP)
- Music On Hold Server (MOH)
- Transcoder (XCODE)
- Annunciator (ANN)

Note

After you configure media resources and if you have not defined any media resource groups, all media resources belong to the default group, and all media resources are available to all Cisco Unified Communications Managers within a given cluster.
Media Resource Group List

A media resource group list provides a prioritized grouping of media resource groups. An application selects the required media resource, such as a music on hold server from the available media resources according to the priority order that you define in a media resource group list. Media resource group lists, which are associated with devices or device pools, provide media resource group redundancy.

Media Resource Group Prerequisites

Ensure that Cisco Unified Communications Manager has media resources to provide services, such as annunciator, transcoding, conferencing, music on hold, and media termination.

Media Resource Group Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Media Resource Groups, on page 439.</td>
<td>Configure media resource group to define a logical groupings of media servers.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign Device to a Media Resource Group, on page 439.</td>
<td>Assign a device to a media resource group. Note: Order of assigning a device is not significant.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Media Resource Group Lists, on page 440.</td>
<td>Create media resource group list to specify a list of prioritized media resource groups. Media resource group lists, which are associated with devices or device pools, provide media resource group redundancy. Note: Order of assigning a device is significant.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Assign Media Resource Group to Media Resource Group List, on page 440.</td>
<td>Assign the newly created media resource group to the media resource group list.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Assign Media Resources to Device or Device Pool, on page 441.</td>
<td>Assign the existing or newly created media resource group list to a device or a device pool.</td>
</tr>
<tr>
<td>Step 6</td>
<td>(Optional) Configure Media Resource Redundancy, on page 441.</td>
<td>Confirm media resources redundancy for a scenario when a media resource fails.</td>
</tr>
</tbody>
</table>
**Configure Media Resource Groups**

A media resource group contains a list of media resources that you want to assign to endpoints, or groups of endpoints.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Media Resources > Media Resource Group**.

**Step 2** Do either of the following:

- Click **Find** and select an existing media resource group.
- Click **Add New** to create a new media resource group.

**Step 3** Configure the fields in the **Media Resource Group Configuration** window. See the online help about the fields and their configuration options.

**Step 4** Enter a **Name** and **Description** for the group.

**Step 5** From **Available Media Resources**, select the resources you want to add to this group, and use the arrows to move the resources to **Selected Media Resources**.

**Step 6** (Optional) To use multicast for Music On Hold audio, check the **Use Multi-cast for MOH Audio** check box.

**Step 7** Click **Save**.

---

**Assign Device to a Media Resource Group**

You can assign devices, such as announciators (ANN), interactive voice responses (IVR), conference bridges (CFB), media termination points (MTP), music on hold (MOH) servers and transcoders to a media resource group. The order in which you assign the devices is not significant.

**Before you begin**

Configure Media Resource Groups, on page 439.

**Procedure**

**Step 1** From the Cisco Unified CM Administration, choose **Media Resources > Media Resource Group**.

**Step 2** To configure an existing media resource group, **Find and List Media Resource Group** window, specify the appropriate filters and click **Find**.

**Step 3** To configure new media resource group, click **Add New**.

**Step 4** From the **Available Media Resources** field, choose the one or multiple devices and click the down-arrow key.

The selected devices appear in the **Selected Media Resources** field.

**Step 5** Click **Save**.
Configure Media Resource Group Lists

Create a prioritized listing of media resource groups. You can assign this list to individual devices or to a device pool.

Procedure

Step 1  From Cisco Unified CM Administration, choose Media Resources > Media Resource Group List.
Step 2  Do either of the following:
   • Click Find and select an existing list.
   • Click Add New and create a new list.
Step 3  Enter a Name for the media resource group list.
Step 4  From Available Media Resource Groups, select the groups you want to add, and use the arrows to move them to Selected Media Resource Groups.
Step 5  Click Save.

Note  For endpoints to use these media resources, you must assign the list to a device pool, gateway port, or to a device.

Assign Media Resource Group to Media Resource Group List

Before you begin

Configure Media Resource Group Lists, on page 440.

Procedure

Step 1  From the Cisco Unified CM Administration, choose Media Resources > Media Resource Group.
Step 2  To configure an existing media resource group, from the Find and List Media Resource Group window, specify the appropriate filters and click Find.
Step 3  From the Available Media Resources list, select one or multiple media resources and click the down arrow key. The selected media resources appear in the Selected Media Resources list.
Step 4  Click Save.

What to do next

Assign Media Resources to Device or Device Pool, on page 441.
Assign Media Resources to Device or Device Pool

Assign media resources to endpoints by associating the prioritized media resource group list to a device pool, or to an individual device.

Procedure

Step 1
From the Cisco Unified CM Administration, choose **Devices > Phone**.
- To add media resources to a device pool, choose **System > Device Pools**.
- To add media resource directly to an endpoint, choose **Device > Phone**.

Step 2
Click **Find** and select the device pool or device to which you want to assign these media resources.

Step 3
From the **Media Resource Group List** drop-down, select a list.

Step 4
Click **Save**.

Step 5
Click **Apply Config to Selected**.

The **Apply Configuration** window appears showing the device name and the applicable configuration changes.

Configure Media Resource Redundancy

Media resource group lists provide media resource redundancy by specifying a prioritized list of media resource groups. An application can select required media resources from among the available ones according to the priority order that is defined in the media resource list.

To configure media resource groups and media resource group lists for redundancy, perform the **Configure Media Resource Groups**, on page 439 and **Media Resource Group List**, on page 438 procedures.

Media Resource Group Interactions and Restrictions

Media Resource Group Interactions

*Table 66: Media Resource Group Interactions*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call processing</td>
<td>Call processing uses a media resource group list in the device level if you select the media resource group list. If a resource is not found, call processing may retrieve it from the default allocation.</td>
</tr>
<tr>
<td></td>
<td>Call processing uses media resource group list in the device pool only if you do not select a media resource group list in the device level. If a resource is not found, call processing may retrieve it from the default allocation.</td>
</tr>
</tbody>
</table>
Interaction Feature

Cisco Unified Communications Manager provides annunciator resources support to a conference bridge if the media resource group list that contains the annunciator is assigned to the device pool where the conference bridge exists. Cisco Unified Communications Manager does not provide annunciator resource support for a conference bridge if the media resource group list is assigned directly to the device that controls the conference.

Video conference

To ensure that only a video conference bridge gets used when a user wants to hold a video conference, add the video conference bridge to a media resource group. Add the media resource group to a media resource group list and assign the media resource group list to the device or device pool that will use the video conference bridge.

## Media Resource Group Restrictions

*Table 67: Media Resource Group Restrictions*

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Deletion of a media resource group</td>
<td>You cannot delete a media resource group that is assigned to a Media Resource Group List.</td>
</tr>
<tr>
<td>Deletion of a transcoder</td>
<td>You cannot delete a transcoder that is assigned to a media resource group.</td>
</tr>
<tr>
<td>Deletion of a media resource</td>
<td>You cannot delete a media resource, such as a conference bridge, that is part of a media resource group unless you first remove the resource from the media resource group or you delete the media resource group that contains the media resource.</td>
</tr>
</tbody>
</table>
Configure Trusted Relay Points

- Trusted Relay Point Overview, on page 443
- Trusted Relay Points Task Flow, on page 444
- Trusted Relay Points Interactions and Restrictions, on page 448

Trusted Relay Point Overview

A Trusted Relay Point (TRP) is an MTP or transcoder that Cisco Unified Communications Manager can insert into the media stream to act as a control point for call media. The TRP can provide further processing on the stream and can ensure that the stream follows a specific path.

When a call requires a trusted relay point, Cisco Unified Communications Manager allocates an MTP or transcoder that has been enabled with TRP functionality.

Configuration

Both MTPs and transcoders can be configured to provide TRP functionality by checking the Trusted Relay Point check box in the Media Termination Point Configuration or Transcoder Configuration window.

You can configure the TRP requirement for individual calls by setting the Use Trusted Relay Point field to On for the following configuration windows:

- Phone Configuration
- Gateway Configuration
- Voicemail Port Configuration
- Trunk Configuration
- CTI Route Point Configuration
- Common Device Configuration
- Universal Device Template Configuration
- Various media resource configurations (Annunciator, IVR, MTPs, Transcoders, Conference Bridges, Music On Hold)
The Trusted Relay Points Task Flow procedure involves configuring trusted relay points (TRPs) for one or multiple devices where media ends and inserting TRPs in Cisco Unified Communications Manager. The steps are as follows:

**Step 1**: Configure TRP in Cisco Unified Communications Manager. Configure media termination point (MTP) so that you can use the device as a trusted relay point.

**Step 2**: Ensure that a device that is configured as a TRP in Cisco Unified Communications Manager has the appropriate network connectivity and configuration between the TRP and any endpoints that are involved in the call.

**Step 3**: Configure transcoder so that you can use the device as a trusted relay point.

**Step 4**: Enable the TRP service parameter to determine whether a call that requires a TRP is allowed to proceed if no TRP resource is available.

---

**Configure Trusted Relay Point for a Device**

You can configure trusted relay points (TRPs) for one or multiple devices where media ends and insert TRPs in Cisco Unified Communications Manager. By configuring the TRP for a device, the device provides further processing on that stream or acts as a method to ensure that the stream follows a specific path.

**Procedure**

**Step 1**: From the Cisco Unified CM Administration, choose **Device > Device Settings > Common Device Configuration**.
Step 2  To configure a trusted relay point for an existing device, from the Find and List Common Device Configurations window, specify the appropriate filters and click Find.

Step 3  To configure trusted relay point for a new device, from the Common Device Configuration window, click Add New.

Step 4  Configure the fields in the Common Device Configuration window. See the online help for more information about the fields and their configuration options.

Step 5  In the Common Device Configuration Information section, click the Use Trusted Relay Point check box.

Step 6  Click Save.

What to do next
Configure Trusted Relay Point for Media Termination Point, on page 445.

Configure Trusted Relay Point for Media Termination Point

You can configure a media termination point (MTP) so that you can use a device as a trusted relay point.

Before you begin
Configure Trusted Relay Point for a Device, on page 444.

Procedure

Step 1  From the Cisco Unified CM Administration, choose Media Resources > Media Termination Point.

Step 2  To configure a trusted relay point for an existing media termination point, from the Find and List Media Termination Points window, specify the appropriate filters and click Find.

Step 3  To configure trusted relay point for a new media termination point, click Add New.

Step 4  Configure the fields on the Media Termination Point Configuration window. See the online help for more information about the fields and their configuration options.

Step 5  In the Media Termination Point Information section, click the Use Trusted Relay Point check box.

Step 6  Click Save.

What to do next
Configure Trusted Relay Point for Transcoder, on page 445.

Configure Trusted Relay Point for Transcoder

You can configure a transcoder so that you can use the device as a trusted relay point.

Before you begin
Configure Trusted Relay Point for Media Termination Point, on page 445.
Enable Trusted Relay Point Service Parameter

You can enable the TRP service parameter to determine whether a call that requires a TRP is allowed to proceed if no TRP resource is available.

Before you begin
Configure Trusted Relay Point for Transcoder, on page 445.

Procedure

Step 1 From the Cisco Unified CM Administration, choose System > Service Parameters. Only Server drop-down list appears.
Step 2 From the Service Parameter Configuration window, choose a server from the Server drop-down list. The Service drop-down list appears.
Step 3 Choose a Cisco Unified Communications Manager server from the Server drop-down list. Based on the selected server and service, the service parameters appear.
Step 4 From the Clusterwide Parameters (Device - General) section, choose True for Fail Call If Trusted Relay Point Allocation Fails drop-down list. See the Related Topics section about the fields and their configuration options.
Step 5 From the Clusterwide Parameters (Device - H323) section, choose True for Fail Call If MTP Allocation Fails drop-down list. See the Related Topics section about the fields and their configuration options.
Step 6 Click Save.

Related Topics
Call Status When MTP and TRP Service Parameters are Selected, on page 447
Call Status When MTP and TRP Service Parameters are Not Selected, on page 447
Call Status When MTP and TRP Service Parameters are Selected

If you check both the Media Termination Point Required and the Use Trusted Relay Point check boxes for an endpoint, Cisco Unified Communications Manager allocates a Media Termination Point (MTP) that is also a Trusted Relay Point (TRP). If the administrator fails to allocate such an MTP or TRP, the call status appears.

The following table shows the call status with the values of the Fail Call If Trusted Relay Point Allocation Fails and the Fail Call if MTP Allocation Fails service parameters when a call fails.

<table>
<thead>
<tr>
<th>Fail Call If TRP Allocation Fails</th>
<th>Fail Call If MTP Allocation Fails</th>
<th>Fail Call?</th>
</tr>
</thead>
<tbody>
<tr>
<td>True</td>
<td>True</td>
<td>Yes</td>
</tr>
<tr>
<td>True</td>
<td>False</td>
<td>Yes</td>
</tr>
<tr>
<td>False</td>
<td>True</td>
<td>Yes, if MTP is required for H.323 endpoint. No, if MTP is required for SIP endpoint.</td>
</tr>
<tr>
<td>False</td>
<td>False</td>
<td>No</td>
</tr>
</tbody>
</table>

Call Status When MTP and TRP Service Parameters are Not Selected

If both the Fail Call If Trusted Relay Point Allocation Fails service parameter and the Fail Call If MTP Allocation Fails service parameter are set to False, the following table shows the call behavior in relationship to the MTP that is required and Use Trusted Relay Point configuration and the resource allocation status.

<table>
<thead>
<tr>
<th>MTP Required</th>
<th>Use TRP</th>
<th>Resource Allocation Status</th>
<th>Call Behavior</th>
</tr>
</thead>
<tbody>
<tr>
<td>Y</td>
<td>Y</td>
<td>TRP allocated</td>
<td>Audio call only because no pass-through support exists.</td>
</tr>
<tr>
<td>Y</td>
<td>Y or N</td>
<td>MTP only</td>
<td>Audio call only. No TRP support.</td>
</tr>
<tr>
<td>Y</td>
<td>Y or N</td>
<td>None allocated</td>
<td>If MTP required is checked for H.323 endpoint, supplementary services will be disabled.</td>
</tr>
<tr>
<td>N</td>
<td>Y</td>
<td>TRP allocated</td>
<td>Audio or video call depends on endpoint capabilities, and call admission control (CAC). Supplementary services still work.</td>
</tr>
<tr>
<td>N</td>
<td>Y</td>
<td>None allocated</td>
<td>Audio or video call. Supplementary services still work, but no TRP support exists.</td>
</tr>
</tbody>
</table>
Trusted Relay Points Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions and Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Resource Reservation Protocol (RSVP)</td>
<td>If RSVP is enabled for the call, Cisco Unified Communications Manager first tries to allocate an RSVP Agent that is also labeled as TRP. Otherwise, another TRP device is inserted between the RSVP Agent and the endpoint.</td>
</tr>
<tr>
<td>Transcoder for call</td>
<td>If you need a transcoder for the call and need to allocate it on the same side as the endpoint that needs TRP, Cisco Unified Communications Manager first tries to allocate a transcoder that is also labeled as TRP. Otherwise, another TRP device is inserted between the transcoder and the endpoint.</td>
</tr>
<tr>
<td>MTP allocation for endpoint</td>
<td>If you check both the Media Termination Point Required check box and the Use Trusted Relay Point check box for an endpoint, Cisco Unified Communications Manager should allocate an MTP that is also a TRP. If the administrator fails to allocate such an MTP or TRP, the call status appears.</td>
</tr>
<tr>
<td>TRP allocation</td>
<td>In most instances, TRP is allocated after users answer the call, so if a call fails due to failure to allocate the TRP, users may receive a fast-busy tone after answering the call. (The SIP outbound leg with MTP required, or H.323 outbound faststart, represents an exception.)</td>
</tr>
<tr>
<td>TRP Insertion for endpoint</td>
<td>Cisco Unified Communications Manager must insert a TRP for the endpoint if you have checked the Use Trusted Relay Point check box for either the endpoint or the device pool that is associated with the device. The call may fail if Cisco Unified Communications Manager fails to allocate a TRP while the Fail Call If Trusted Relay Point Allocation Fails service parameter is set to True.</td>
</tr>
</tbody>
</table>

Table 68: Trusted Relay Points Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Insertion of trusted relay point for an endpoint</td>
<td>Cisco Unified Communications Manager must insert a TRP for the endpoint if you have checked the Use Trusted Relay Point check box for either the endpoint or the device pool that is associated with the device. The call may fail if Cisco Unified Communications Manager fails to allocate a TRP while the Fail Call If Trusted Relay Point Allocation Fails service parameter is set to True.</td>
</tr>
</tbody>
</table>
### Trusted Relay Points Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allocation of media termination point for an endpoint</td>
<td>If you check both the <strong>Media Termination Point Required</strong> check box and the <strong>Use Trusted Relay Point</strong> check box for an endpoint, Cisco Unified Communications Manager should allocate an MTP that is also a TRP. If the administrator fails to allocate such an MTP or TRP, the call status appears.</td>
</tr>
<tr>
<td>Allocation of trusted relay point</td>
<td>In most instances, TRP is allocated after users answer the call, so if a call fails due to failure to allocate the TRP, users may receive fast-busy tone after answering the call. (The SIP outbound leg with MTP required, or H.323 outbound faststart, represents an exception.)</td>
</tr>
</tbody>
</table>
Trusted Relay Points Restrictions
Configure Annunciator

- Annunciator Overview, on page 451
- Annunciator Configuration Task Flow, on page 453

Annunciator Overview

An annunciator is an SCCP software device that runs on Cisco Unified Communications Manager and which allows you to send prerecorded messages and tones to Cisco IP Phones and gateways. The annunciator is activated on a cluster node by turning on the Cisco IP Voice Media Streaming service on that node. Features such as MLPP, SIP trunks, IOS gateways, and software conference bridges rely on the annunciator to send the predefined message to the phone or gateway via a one-way media stream. In addition:

- Both IPv4 and IPV6 are supported. The annunciator is configured automatically in dual mode when the system's platform is configured for IPv6 and the IPv6 enterprise parameter is enabled.
- SRTP is supported

Annunciator Scalability

By default, an annunciator supports 48 simultaneous media streams. You can add capacity by activating the annunciator on additional nodes or by changing the default number of annunciator media streams via the Call Count service parameter. However, it's not recommended to increase this value on a node unless the Cisco CallManager service is deactivated on that node.

If the annunciator runs on a dedicated subscriber node where the Cisco CallManager service does not run, the annunciator can support up to 255 simultaneous announcement streams. If the dedicated subscriber node meets the OVA virtual machine configuration for 10,000 users, the annunciator can support up to 400 simultaneous announcement streams.

Caution

We recommend that you do not activate the annunciator on Unified Communications Manager nodes that have a high call-processing load.

Annunciator with Conference Bridge

The Annunciator is available to a conference bridge under the following conditions:
• If the media resource group list that contains the annunciator is assigned to the device pool where the conference bridge exists.

• If the annunciator is configured as the default media resource.

The annunciator is not available to a conference bridge if the media resource group list is assigned directly to the device that controls the conference.

Each conference supports only one announcement. If the system requests another announcement while the current announcement is playing, the new announcement preempts the one that is playing.

**Default Annunciator Announcements and Tones**

Cisco Unified Communications Manager automatically provides a set of prerecorded annunciator announcements when you activate the Cisco IP Media Streaming Application service. An announcement or a tone is played for the following conditions:

• Announcement — Played for devices that are configured for Cisco Multilevel Precedence and Preemption.

• Barge tone — Heard before a participant joins an ad hoc conference.

• Ring back tone — When you transfer a call over the PSTN through an IOS gateway, the annunciator plays the tone because the gateway cannot play the tone when the call is active.

• Ring back tone — When you transfer calls over an H.323 intercluster trunk, a tone is played.

• Ring back tone — When you transfer calls to the SIP client from a phone that is running SCCP, a tone is played.

You cannot change the default prerecorded annunciator announcements or add additional announcements. Localization of the announcement is supported if the Cisco Unified Communications Manager Locale Installer is installed and the locale settings are configured for the Cisco Unified IP Phone or device pool. For information about the Locale Installer and the files to install for user and (combined) network locales, see *Installing Cisco Unified Communications Manager*. To download the locale installer, see the support pages at [www.cisco.com](http://www.cisco.com).

**Table 69: Prerecorded Annunciator Announcements**

<table>
<thead>
<tr>
<th>Condition</th>
<th>Announcement</th>
</tr>
</thead>
<tbody>
<tr>
<td>An equal or higher precedence call is in progress.</td>
<td>Precedence access limitation has prevented the completion of your call. Please hang up and try again. This is a recording.</td>
</tr>
<tr>
<td>A precedence access limitation exists.</td>
<td>Precedence access limitation has prevented the completion of your call. Please hang up and try again. This is a recording.</td>
</tr>
<tr>
<td>Someone attempted an unauthorized precedence level.</td>
<td>The precedence used is not authorized for your line. Please use an authorized precedence or ask your operator for assistance. This is a recording.</td>
</tr>
<tr>
<td>The call appears busy, or the administrator did not configure the directory number for call waiting or preemption.</td>
<td>The number you have dialed is busy and not equipped for call waiting or preemption. Please hang up and try again. This is a recording.</td>
</tr>
</tbody>
</table>
Announcement

The system cannot complete the call. Your call cannot be completed as dialed. Please consult your directory and call again or ask your operator for assistance. This is a recording.

A service interruption occurred. A service disruption has prevented the completion of your call. In case of emergency call your operator. This is a recording.

The following table lists the tones that the annunciator supports.

Table 70: Tone Description

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Busy tone</td>
<td>A busy tone is heard when the dialed number is busy.</td>
</tr>
<tr>
<td>Barge tone</td>
<td>A conference barge-in tone is heard before the participant joins an ad hoc conference.</td>
</tr>
<tr>
<td>Ring back tone</td>
<td>An alert tone is heard for the following scenarios:</td>
</tr>
<tr>
<td></td>
<td>• When you transfer a call over the PSTN through an IOS gateway.</td>
</tr>
<tr>
<td></td>
<td>• When you transfer a call over an H.323 intercluster trunk.</td>
</tr>
<tr>
<td></td>
<td>• When you transfer a call to the SIP client from an SCCP phone.</td>
</tr>
</tbody>
</table>

Annunciator Use With Conference Bridges

The Annunciator is available to a conference bridge under the following conditions:

• If the media resource group list that contains the annunciator is assigned to the device pool where the conference bridge exists.
• If the annunciator is configured as the default media resource.

The annunciator is not available to a conference bridge if the media resource group list is assigned directly to the device that controls the conference.

Each conference supports only one announcement. If the system requests another announcement while the current announcement is playing, the new announcement preempts the one that is playing.

Related Topics

Media Resource Group Overview

Annunciator Configuration Task Flow

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Activate the Annunciator, on page 454</td>
<td>Activate the Cisco IP Voice Media Streaming Application service on the node to activate the</td>
</tr>
</tbody>
</table>
Activate the Annunciator

Activate only one Cisco IP Voice Media Streaming Application service for each annunciator device in the cluster.

### Caution

We recommend that you do not activate the annunciator on Cisco Unified Communications Manager nodes that have a high call-processing load.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>Required: Media Resource Group Task Flow, on page 438</td>
<td>Add the annunciator to media resource groups and lists to manage your media resources using Cisco Unified Communications Manager Administration. You can see which media resource groups have annunciators from the <strong>Dependency Records Summary</strong> window.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td></td>
</tr>
<tr>
<td>Configure Device Pools, on page 54</td>
<td>Add the media resource group that includes the annunciator to a device pool using Cisco Unified Communications Manager Administration. Repeat this step for each annunciator. Each annunciator must belong to a device pool.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td></td>
</tr>
<tr>
<td>(Optional) Change the Default Number of Media Streams, on page 455</td>
<td>You can change the default number of media streams for an annunciator.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td></td>
</tr>
<tr>
<td>(Optional) Override the Annunciator Security Mode, on page 455</td>
<td>When Cisco Unified Communications Manager is configured in a secured deployment, the media streaming between the annunciator and security enabled devices is automatically encrypted using the Secure Real-Time Protocol (SRTP). You can override the annunciator security settings so that media streamed from the secured annunciator is not encrypted.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td></td>
</tr>
<tr>
<td>(Optional) View List of Media Resource Groups That Have Annunciators, on page 456</td>
<td>You can see which media resource groups use the annunciator device.</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td></td>
</tr>
<tr>
<td>(Optional) Configure Annunciator for Conference Bridges, on page 457</td>
<td>You can use the annunciator with conference bridges when the annunciator and conference bridge belong to the same device pool.</td>
</tr>
</tbody>
</table>
### Configure Media Resources

#### Change the Default Number of Media Streams

An announciator supports 48 simultaneous media streams by default. You can change the default number of media streams using the announciator service parameter; however, we recommend that you do not exceed 48 annunciator streams on a node.

**Before you begin**

- Activate the Announcer, on page 454

**Procedure**

1. From Cisco Unified Communications Manager Administration, choose System > Service Parameters.
2. In the Service Parameters Configuration window, select the server and then select the service called Cisco IP Voice Media Streaming App.
3. In the Service Parameter Configuration window, enter the number of simultaneous media streams in the Call Count field of the Announcer (ANN) Parameters section, and then click Save.

When you update the announciator, the changes automatically occur when the announciator is idle and no active announcements are playing.

**What to do next**

- Override the Announcer Security Mode, on page 455

### Override the Announcer Security Mode

When the enterprise parameter called Cluster Security Mode is set to 1 (mixed mode), annunciator devices are automatically enable for security. The annunciator registers as a secured SRTP device on Cisco Unified Communications Manager nodes that have Secure Real-Time Protocol (SRTP) enabled. A locked icon appears on SRTP capable devices. Announcements from a secured annunciator are encrypted if the receiving device is also SRTP capable; otherwise, unsecured announcements and tones are sent.
You can override the annunciator security mode using the service parameter called Make Annunciator Non-secure when Cluster Security is Mixed. When the annunciator security mode is overridden, an unencrypted announcement is played even if the receiving device is SRTP capable.

**Before you begin**

*Change the Default Number of Media Streams, on page 455*

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Required: From Cisco Unified Communications Manager Administration, choose <strong>System &gt; Service Parameters</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Select the node in the <strong>Server</strong> field.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Select <strong>Cisco Unified IP Voice Media Streaming Application</strong> in the <strong>Service</strong> field.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Set <strong>Make Annunciator Non-secure when Cluster Security is Mixed</strong> to <strong>True</strong>, and then click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**Tip** Click **Advanced** if you do not see the **Make Annunciator Non-secure when Cluster Security is Mixed** parameter.

**What to do next**

*Configure Annunciator for Conference Bridges, on page 457*

---

**View List of Media Resource Groups That Have Annunciators**

View the **Dependency Records Summary** window to see which media resource groups use the annunciator device.

**Before you begin**

*Override the Annunciator Security Mode, on page 455*

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified CM Administration, choose <strong>Media Resources &gt; Annunciator</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Select the annunciator that is set up for your system.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>From the <strong>Related Links</strong> drop-down list box, choose <strong>Dependency Records</strong> and click <strong>Go</strong>. The <strong>Dependency Records Summary</strong> window displays the media resource groups that use the annunciator device.</td>
</tr>
</tbody>
</table>

**What to do next**

*Configure Annunciator for Conference Bridges, on page 457*
Configure Annunciator for Conference Bridges

You can use the annunciator with conference bridges.

Before you begin

View List of Media Resource Groups That Have Annunciators, on page 456

Procedure

Step 1
Add the annunciator to a media resource group list.

Step 2
Assign the media resource group list that contains the annunciator to the device pool of the conference bridge to make the annunciator available to all devices in the cluster.

Related Topics

- Media Resource Group Overview
- Media Resource Group Task Flow, on page 438
- Device Pools
- Configure a Basic Device Pool
Configure Annunciator for Conference Bridges
CHAPTER 61

Configure Interactive Voice Response

- Interactive Voice Response Overview, on page 459
- Default IVR Announcements and Tones, on page 459
- Interactive Voice Response Restrictions, on page 460
- Interactive Voice Response Configuration Task Flow, on page 461

Interactive Voice Response Overview

The Interactive Voice Response (IVR) device enables Cisco Unified Communications Manager to play prerecorded feature announcements (.wav files) to devices such as Cisco Unified IP Phones and Gateways. These announcements play on devices that use features which require IVR announcements, like Conference Now.

When you add a node, an IVR device is automatically added to that node. The IVR device remains inactive until the Cisco IP Voice Media Streaming Application service is activated on that node.

An IVR supports 48 simultaneous callers by default. You can change the number of IVR callers using the Cisco IP Voice Media Streaming Application service parameter. However, we recommend that you do not exceed 48 IVR callers on a node. You can configure the number of callers for IVR based on expected simultaneous calls to IVR for joining Conference Now.

Caution

Do not activate the IVR device on Cisco Unified Communications Manager nodes that have a high call-processing load.

Default IVR Announcements and Tones

Cisco Unified Communications Manager automatically provides a set of prerecorded Interactive Voice Response (IVR) announcements when you activate the Cisco IP Media Streaming Application service. You can replace the default prerecorded IVR announcements. An announcement is played for the following conditions:
Table 71: Prerecorded IVR Announcements

<table>
<thead>
<tr>
<th>Announcement</th>
<th>Condition</th>
</tr>
</thead>
<tbody>
<tr>
<td>ConferenceNowAccessCodeFailed Announcement</td>
<td>Plays when an attendee enters the wrong access code to join Conference Now after exceeding the maximum number of attempts.</td>
</tr>
<tr>
<td>ConferenceNowAccessCodeInvalid Announcement</td>
<td>Plays when an attendee enters the wrong access code.</td>
</tr>
<tr>
<td>ConferenceNowCFBFailed Announcement</td>
<td>Plays when the conference bridge capacity limit is exceeded while initiating Conference Now.</td>
</tr>
<tr>
<td>ConferenceNowEnterAccessCode Announcement</td>
<td>Plays when an attendee joins Conference Now and the host sets an attendee access code.</td>
</tr>
<tr>
<td>ConferenceNowEnterPIN Announcement</td>
<td>Plays when a host or attendee tries to join a meeting.</td>
</tr>
<tr>
<td>ConferenceNowFailedPIN Announcement</td>
<td>Plays after the host exceeds the maximum number of attempts to enter a correct PIN.</td>
</tr>
<tr>
<td>ConferenceNowGreeting Announcement</td>
<td>Plays a greeting prompt for Conference Now.</td>
</tr>
<tr>
<td>ConferenceNowInvalidPIN Announcement</td>
<td>Plays when the host enters a wrong PIN.</td>
</tr>
<tr>
<td>ConferenceNowNumberFailed Announcement</td>
<td>Plays when a host or attendee enters the wrong meeting number after exceeding the maximum number of attempts.</td>
</tr>
<tr>
<td>ConferenceNowNumberInvalid Announcement</td>
<td>Plays when a host or attendee enters a wrong meeting number.</td>
</tr>
</tbody>
</table>

Interactive Voice Response Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Load Balancing</td>
<td>The Interactive Voice Response (IVR) uses Real-Time Protocol (RTP) streams through a common media device driver. This device driver is also used by other software media devices provided by the Cisco IP Voice Media Streaming Application services such as Music On Hold (MOH), Software Media Termination Point (MTP), Software Conference Bridge (CFB), and Annunciator. Configuring a larger call volume affects the system performance. This also impacts call processing if the Call Manager service is active on the same server node.</td>
</tr>
<tr>
<td>DTMF Digits</td>
<td>The IVR supports only Out-Of-Band (OOB) DTMF digit collection method. If there is a DTMF capability mismatch between the calling device and the IVR, an MTP will be allocated.</td>
</tr>
</tbody>
</table>
Restriction Feature

The IVR only supports codec G.711 (a-law and μ-law), G.729, and Wide Band 256k. If there is a codec mismatch between the calling device and the IVR, a transcoder will be allocated.

### Interactive Voice Response Configuration Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td><strong>Activate the Interactive Voice Response, on page 461</strong></td>
<td>Activate the Cisco IP Voice Media Streaming Application service on the node to activate the IVR for that node. Activate only one Cisco IP Voice Media Streaming Application service for each IVR device in the cluster.</td>
</tr>
<tr>
<td>Step 2</td>
<td><strong>Required: View List of Media Resource Groups That Have IVR, on page 462</strong></td>
<td>Add the IVR to media resource groups and lists to manage your media resources using Cisco Unified Communications Manager Administration.</td>
</tr>
<tr>
<td>Step 3</td>
<td><strong>(Optional) Change the Default Number of Media Streams, on page 455</strong></td>
<td>You can change the default number of media streams for an IVR.</td>
</tr>
</tbody>
</table>

### Activate the Interactive Voice Response

Activate one or more Cisco IP Voice Media Streaming Application service for each node to have Interactive Voice Response (IVR) device registered in the cluster.

**Caution**

Do not activate the IVR on Cisco Unified Communications Manager nodes that have a high call-processing load.

**Procedure**

**Step 1**  
From the Cisco Unified Serviceability GUI, choose **Tools > Activation**. The **Service Activation** window appears.

**Step 2**  
Select the node in the **Server** field and click **Go**.

**Step 3**  
Check the **Cisco IP Voice Media Streaming Application** check box, and then click **Save**.
View List of Media Resource Groups That Have IVR

Procedure

Step 1  From Cisco Unified CM Administration, choose Media Resources > Interactive Voice Response (IVR). The Find and List Interactive Voice Response (IVR) window is displayed.

Step 2  From the Find and List Interactive Voice Response (IVR) where window, click Find. A list of IVRs that are available on Cisco Unified Communications Manager is displayed.

Step 3  Choose the IVR on which you want to see the associated list of media resource groups.

Step 4  Choose Dependency Records from the Related Links drop-down list and click Go.

If the dependency records are not enabled for the system, the Dependency Records Summary window displays a message.

IVR Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Server</td>
<td>Displays the preconfigured server (servers are added at installation), by default.</td>
</tr>
<tr>
<td>Name</td>
<td>Designates the name that is used when the device registers with the Cisco Unified Communications Manager. Enter a name of up to 15 alphanumeric characters (you can use periods, dashes, and underscores).</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description of up to 128 alphanumeric characters (you can use periods, dashes, and underscores). Default uses the server name, which includes the prefix IVR_.</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Choose Default or choose a device pool from the drop-down list of configured device pools.</td>
</tr>
<tr>
<td>Location</td>
<td>Use locations to implement call admission control (CAC) in a centralized call-processing system. CAC allows you to regulate audio quality and video availability by limiting the bandwidth that is available for audio and video calls over links between locations. The location specifies the total bandwidth that is available for calls to and from this location. From the drop-down list, choose the appropriate location for this IVR. A location setting of Hub_None means that the locations feature does not keep track of the bandwidth that this IVR consumes. A location setting of Phantom specifies a location that enables successful CAC across intercluster trunks that use H.323 protocol or SIP. To configure a new location, use the System &gt; Location menu option. For details on setting up a location-based CAC across intercluster trunks, see the System Configuration Guide for Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Use Trusted Relay Point  | From the drop-down list, enable or disable whether Unified Communications Manager inserts a trusted relay point (TRP) device with this media endpoint. Choose one of the following values:  
  • Off—Choose this value to disable the use of a TRP with this device.  
  • On—Choose this value to enable the use of a TRP with this device.  

A Trusted Relay Point (TRP) device designates an MTP or transcoder device that is labeled as Trusted Relay Point.  
Unified Communications Manager places the TRP closest to the associated endpoint device if more than one resource is needed for the endpoint (for example, a transcoder or RSVPAgent).  
If both TRP and MTP are required for the endpoint, TRP gets used as the required MTP.  
If both TRP and RSVPAgent are needed for the endpoint, Unified Communications Manager searches for an RSVPAgent that can also be used as a TRP.  
If both TRP and transcoder are needed for the endpoint, Cisco Unified Communications Manager searches for a transcoder that is also designated as a TRP.

Change IVR Parameters

Procedure

Step 1  
From Cisco Unified Communications Manager Administration, choose System > Service Parameters. The Service Parameters Configuration window appears.

Step 2  
Select the server and then select the service called Cisco IP Voice Media Streaming App. The Service Parameter Configuration window appears.

Step 3  
Enter the number of simultaneous media streams in the Call Count field of the Interactive Voice Response (IVR) Parameters section, and then click Save.

When you update the IVR, the changes automatically occur when the IVR is idle and no active announcements are playing.
Configure Media Resources

Change IVR Parameters
CHAPTER 62

Configure Video On Hold Server

- Video On Hold Overview, on page 465
- Video on Hold Configuration Task Flow, on page 466
- Video On Hold Interactions, on page 467

Video On Hold Overview

The video on hold feature is for video contact centres where customers calling into the video contact centre are able to watch a specific video after initial consultation with the agent at the contact centre. In this case, the agent selects the video stream that is played to the customer while the customer is on hold.

The video on hold server is a media content server that can stream audio and video content when directed by Cisco Unified Communications Manager. The media content server is an external device that can store and stream audio and video content under Unified Communications Manager control using SIP as the signal protocol. The media content server is capable of providing hi-definition video content at 1080p, 720p, or lower resolutions such as 360p. Cisco MediaSense is used as the media content server.

In addition to the video contact centre, video on hold can be deployed within any enterprise if the deployment requires a generic video on hold capability. You can configure a Default Video Content Identifier for the video on hold server which identifies the video stream that is played to the user that is on hold.

Note

In a Unified Contact Center with Customer Voice Portal (CVP) post-routed deployment, to get video on hold functionality, you must allocate video on hold resources in the SIP trunk that runs between Unified Communications Manager and CVP.
Video on Hold Configuration Task Flow

Before you begin

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Create a SIP Trunk to Cisco MediaSense Server, on page 466</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure a Video On Hold Server, on page 467</td>
</tr>
</tbody>
</table>

Create a SIP Trunk to Cisco MediaSense Server

You must configure Unified Communications Manager with a SIP trunk to a Cisco MediaSense cluster. The SIP trunk to the Cisco MediaSense server contains the IP addresses of the Cisco MediaSense nodes. The Unified Communications Manager SIP trunk supports up to 16 destination IP addresses.

Note
Cisco MediaSense cluster should have two or more nodes for redundancy and scalability purposes.
You configure the SIP trunk with default configuration. Other configurations on the SIP trunk are not supported for the video on hold feature.

Procedure

Step 1 | In Cisco Unified CM Administration, choose Device > Trunk. |
Step 2 | Click Add New. |
Step 3 | From the Trunk Type drop-down list, choose SIP Trunk. |
Step 4 | From the Device Protocol drop-down list, ensure that SIP is entered as the protocol and click Next. |
Step 5 | In the Device Information area, enter the following fields: |
| | • Device Name—Enter a trunk name. |
| | • Description—Enter a trunk description. |
| | • Device Pool—Choose the appropriate device pool for the SIP trunk. |
| | • Location—Choose the appropriate location for this trunk. |
Step 6 | In the SIP Information area, enter the following fields: |
| | • Destination Address—Enter the IP address of the Cisco MediaSense server. You can specify multiple IP addresses. |
Configure a Video On Hold Server

The SIP trunk for the video on hold server points to the Cisco MediaSense server and the default content identifier points to a stream ID that exists on the MediaSense server. The content identifier can be any alphanumeric string.

Before you begin

Create a SIP Trunk to Cisco MediaSense Server, on page 466

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose Media Resources > Video on Hold Server.
Step 2 Click Add New to set up a new video on hold server.
Step 3 Enter the name of the video on hold server.
Step 4 Enter a description for the server.
Step 5 Enter an alphanumeric string for the Default Video Content Identifier.
Step 6 Select the SIP trunk to be used from the drop-down list. If a new SIP trunk needs to be created, click the Create SIP Trunk button
Step 7 Click Save.

Video On Hold Interactions

For the Enhanced Location Call Admission Control feature, the Cisco MediaSense servers can be co-located in a Unified Communications Manager cluster (the Cisco MediaSense cluster is directly connected to the cluster where the holding party is registered). In that case, the Unified Communications Manager cluster is responsible for deducting the bandwidth between the location of the party on hold and the Cisco MediaSense location. Since video on hold interactions make use of 720p or 1080p video streams, it is important to take
the bandwidth usage into account before allowing new sessions in order to maintain video quality of existing sessions.
CHAPTER 63

Configure Announcements

- Announcements Overview, on page 469
- Announcements Configuration Task Flow, on page 470

Announcements Overview

In Cisco Unified Communications Manager Administration, use the Menu Resources > Announcements menu path to configure announcements. There are two classifications of announcements:

- System Announcements—Pre-defined announcements that are used in normal call processing or provided as sample feature announcements.
- Feature Announcements—Used by features such as Music on Hold (MOH), Hunt Pilots with Call Queuing or External Call Control. You can customize your own feature announcements by uploading Cisco-provided audio files or uploading custom .wav files. Upload all custom announcement .wav files to all servers in the cluster.

Note

You can hear custom announcements such as warning or reorder tones if you are connected through a trunk or gateway. However, you cannot hear custom announcements on calls between two IP phones or IP phones and Jabber clients.

Formats

The recommended format for announcements includes the following specifications:

- 16-bit PCM wav file
- Stereo or mono
- Sample rates of 48 kHz, 44.1 kHz, 32 kHz, 16 kHz, or 8 kHz

Default Announcements

You can upload custom announcement .wav files or change the Cisco-provided file for a system announcement. However, you cannot change the announcement identifier. For example, the System announcement
Announcements Configuration Task Flow

(VCA_00121) is played when a caller dials an invalid number. This is commonly known as the vacant call announcement.

Table 72: Announcements in the Find and List Announcements Window

<table>
<thead>
<tr>
<th>Announcement Identifier</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gone_00126</td>
<td>System: Gone</td>
</tr>
<tr>
<td>MLPP-BNEA_00123</td>
<td>System: MLPP Busy not equipped</td>
</tr>
<tr>
<td>MLPP-BPA_00122</td>
<td>System: MLPP Higher precedence</td>
</tr>
<tr>
<td>MLPP-ICA_00120</td>
<td>System: MLPP Service disruption</td>
</tr>
<tr>
<td>MLPP-PALA_00119</td>
<td>System: MLPP Precedence access limit</td>
</tr>
<tr>
<td>MLPP-UPA_00124</td>
<td>System: MLPP Unauthorized precedence</td>
</tr>
<tr>
<td>Mobility_VMA</td>
<td>Please press 1 to be connected</td>
</tr>
<tr>
<td>MonitoringWarning_00055</td>
<td>System: Monitoring or Recording</td>
</tr>
<tr>
<td>RecordingWarning_00038</td>
<td>System: Recording</td>
</tr>
<tr>
<td>TemporaryUnavailable_00125</td>
<td>System: Temporary unavailable</td>
</tr>
<tr>
<td>VCA_00121</td>
<td>System: Vacant number / invalid number dialed</td>
</tr>
<tr>
<td>Wait_In_Queue_Sample</td>
<td>Built-in: Sample queued caller periodic announcement</td>
</tr>
<tr>
<td>Welcome_Greeting_Sample</td>
<td>Built-in: Sample caller greeting</td>
</tr>
</tbody>
</table>

Announcements Configuration Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configure Announcement, on page 471.</td>
<td>Configure an announcement that you can use with features, such as Music On Hold (MoH) along with Hunt Pilot call queuing or External Call Control.</td>
</tr>
<tr>
<td>Upload a Customized Announcement, on page 471.</td>
<td>Upload custom announcement .wav files or change the Cisco-provided file for a system announcement. However, you cannot change the announcement identifier. The customized announcements are underlined with a hyperlink and appear in the <strong>Find and List Announcements</strong> window of Cisco Unified Communications Manager.</td>
</tr>
</tbody>
</table>
Configure Announcement

You can configure an announcement that you can use as a system announcement or as a feature announcement. A system announcement is used for call processing or for the use of sample feature announcements whereas a feature announcement is used for specific features, such as music on hold (MOH) in association with hunt pilot call queuing or external call control.

You can modify an existing announcement or configure a new announcement in Cisco Unified Communications Manager.

Procedure

Step 1 From Cisco Unified CM Administration, choose Media Resources > Announcement.

Step 2 Do one of the following:
   • Click Find and select an existing announcement to edit.
   • Click Add New to add a new announcement.

Step 3 Configure the fields in the Announcement Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 4 Click Save.

Upload a Customized Announcement

You can modify a default announcement with an uploaded custom .wav file with a different announcement. When you import an audio source file, Unified Communications Manager processes the file and converts the file to the proper formats for use by the music on hold (MOH) server.

Announcements are specific to the locale (language). If your installation is using more than one language locale, you have to record each custom announcement each language as a separate .wav file and upload with the correct locale assignment. This task also requires that the correct locale package is installed on each server before uploading custom announcement .wav files for languages other than United States English.

Similar to MOH audio source files, the recommended format for announcements includes the following specifications:
   • 16-bit PCM .wav file
   • Stereo or mono
   • Sample rates of 48 kHz, 44.1 kHz, 32 kHz, 16 kHz, or 8 kHz

You cannot update announcements that are not hyperlinked in the Find and List Announcements window in Unified Communications Manager. You can add customized announcements for Cisco-provided announcements that are underlined with a hyperlink in this window. For example, MLPP-ICA_00120 and MonitoringWarning_00055.
Upload a Customized Announcement

Configure Media Resources

Procedure

Step 1  From Cisco Unified CM Administration, choose **Media Resources > Announcement**.

Step 2  From the **Find and List Announcements** window, enter search criteria, click **Find**, and click the hyperlink for the announcement from the resulting list.

Step 3  From the **Announcement Configuration** window, click **Upload File**.

Step 4  From the **Upload File** pop-up window, choose the locale, enter the filename and browse to select the .wav file, and click **Upload File**.

The upload process begins and the status is updated after the processing is complete. Select **Close** to close the **Upload File** window.

Step 5  (Optional) If you want Unified Communications Manager to play the customized announcement instead of playing the Cisco-provided announcement, check the **Enable** check box appears in the **Announcement by Locale** pane in the **Announcements Configuration** window.

If the **Enable** check box is unchecked, Unified Communications Manager plays the Cisco-provided announcement.

Step 6  Click **Save**.

What to do next

Upload the announcement on each node in the cluster as the announcement files are not propagated between servers in a cluster. Browse for Cisco Unified Communications Manager Administration on each server in the cluster and repeat the upload process.
Configure Conference Bridges

Conference Bridges Overview

Conference bridge for Cisco Unified Communications Manager is a software or hardware application that is designed to allow both ad hoc and meet-me voice conferencing. Additional conference bridge types support other types of conferences, including video conferences. Each conference bridge can host several simultaneous, multiparty conferences. Both hardware and software conference bridges can be active at the same time.

Software and hardware conference bridges differ in the number of streams and the types of codec that they support. When you add a new server, the system automatically adds software conference bridges.

Note

When Cisco Unified Communications Manager server is created, the Conference Bridge Software is also created automatically and it cannot be deleted. You cannot add Conference Bridge Software to Cisco Unified Communications Manager Administration.

Conference Bridge Types

The following conference bridge types are available in Cisco Unified Communications Manager Administration.
## Table 73: Conference Bridge Types

<table>
<thead>
<tr>
<th>Conference Bridge Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Conference Bridge Hardware</td>
<td>This type supports the Cisco Catalyst 4000 and 6000 Voice Gateway Modules and the following number of conference sessions:</td>
</tr>
<tr>
<td></td>
<td><strong>Cisco Catalyst 6000</strong></td>
</tr>
<tr>
<td></td>
<td>• G.711 or G.729a conference - 32 participants per port; six participants maximum per conference; 256 total participants per module; 10 bridges with three participants.</td>
</tr>
<tr>
<td></td>
<td>• GSM - 24 participants per port; six participants maximum per conference; 192 total participants per module.</td>
</tr>
<tr>
<td></td>
<td><strong>Cisco Catalyst 4000</strong></td>
</tr>
<tr>
<td></td>
<td>G.711 conference only - 24 conference participants; maximum of four conferences with six participants each.</td>
</tr>
<tr>
<td>Cisco Conference Bridge Software</td>
<td>Software conference devices support G.711 codecs by default.</td>
</tr>
<tr>
<td></td>
<td>The maximum number of callers for this type equals 256. With a setting of 256, the software conference bridge can support 64 conference sessions of 4 parties each. The maximum number of caller parties in a conference session is specified via the Maximum Ad Hoc Conference and Maximum MeetMe Conference Unicast service parameters.</td>
</tr>
<tr>
<td></td>
<td><strong>Caution</strong></td>
</tr>
<tr>
<td></td>
<td>This type of conference bridge (SW Conference Bridge) is a simplified implementation. It does not identify parties that are silent and uses a simple summing algorithm which may cause audio quality and low volume levels for the conference when there is a large number of participants.</td>
</tr>
<tr>
<td>Cisco IOS Conference Bridge</td>
<td>• Uses the NM-HDV or NM-HDV-FARM network modules.</td>
</tr>
<tr>
<td></td>
<td>• G.711 a/mu-law, G.729, G.729a, G.729b, and G.729ab participants can join in a single conference call</td>
</tr>
<tr>
<td></td>
<td>• Up to six parties can join in a single conference call</td>
</tr>
<tr>
<td></td>
<td>Cisco Unified Communications Manager assigns conference resources to calls on a dynamic basis.</td>
</tr>
<tr>
<td></td>
<td>For more information about Cisco IOS Conferencing and Transcoding for Voice Gateway Routers, see the Cisco IOS documentation that you received with this product.</td>
</tr>
<tr>
<td>Conference Bridge Type</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Cisco IOS Enhanced Bridges                     | • Uses the onboard Cisco Packet Voice/Fax Digital Signal Processor Modules (PVDM2) on the Cisco 2800 and 3800 series voice gateway routers or uses the NM-HD or NM-HDV2 network modules.  

• G.711 a-law/mu-law, G.729, G.729a, G.729b, G.729ab, GSM FR, and GSM EFR participants can join in a single conference  

• Up to eight parties can join in a single call.  

**Note**  
With ISR4000 router and any of the SM-X-PVDM-3000/SM-X-PVDM-2000/SM-X-PVDM-1000/SM-X-PVDM-500, each Conference Bridge Profile can register up to a maximum of 512 sessions, due to the Unified Communications Manager 4096 maximum stream limitation.  
Cisco Unified Communications Manager assigns conference resources to calls on a dynamic basis.  
For more information about Cisco IOS Enhanced Conferencing and Transcoding for Voice Gateway Routers, see the Cisco IOS documentation that you received with this product.  
This conference bridge type supports SRTP media encryption with AES_CM_128_HMAC_SHA1_80 for supported SIP phones where an ISR 4000 series gateway is deployed. SCCP phones and non-supported SIP phones fall back to AES_CM_128_HMAC_SHA1_32 encryption.  
**Note**  
Make sure that the gateway load supports the cipher. Please review your gateway documentation for support details.  

| Cisco Conference Bridge (WS-SVC-CMM)          | This conference bridge type supports the Cisco Catalyst 6500 series and Cisco 7600 series Communication Media Module (CMM).  

It supports up to eight parties per conference and up to 64 conferences per port adapter. This conference bridge type supports the following codecs: This conference bridge type supports ad hoc conferencing.  

• G.711 a-law/mu-law  

• G.729 annex A and annex B  

• G.723.1  


**Note**  
This conference bridge type supports SRTP media encryption with AES_CM_128_HMAC_SHA1_80 for supported SIP phones where an ISR 4000 series gateway is deployed. SCCP phones and non-supported SIP phones fall back to AES_CM_128_HMAC_SHA1_32 encryption.  
Make sure that the gateway load supports the cipher. Please review your gateway documentation for support details.  

**Conference Bridge Types**
<table>
<thead>
<tr>
<th>Conference Bridge Type</th>
<th>Description</th>
</tr>
</thead>
</table>
| Cisco TelePresence MCU         | Cisco TelePresence MCU is a set of hardware conference bridges for Cisco Unified Communications Manager.  
The Cisco TelePresence MCU is a high-definition (HD) multipoint video conferencing bridge. It delivers up to 1080p at 30 frames per second, full continuous presence for all conferences, full transcoding, and is ideal for mixed HD endpoint environments.  
The Cisco TelePresence MCU supports SIP as the signaling call control protocol. It has a built-in Web Server that allows for complete configuration, control, and monitoring of the system and conferences. The Cisco TelePresence MCU provides XML management API over HTTP.  
Cisco TelePresence MCU allows both ad hoc and meet-me voice and video conferencing. Each conference bridge can host several simultaneous, multiparty conferences.  
Cisco Unified Communications Manager supports presentation sharing with the Binary Floor Control Protocol (BFCP) between Unified Communications Manager and a Cisco TelePresence MCU.  
Cisco TelePresence MCU must be configured in Port Reservation mode. For more information, consult the Cisco TelePresence MCU Configuration Guide.  
**Note** Cisco TelePresence MCU does not support a common out-of-band DTMF method. Under the default setting, Cisco Unified Communications Manager will not require a Media Termination Point (MTP). However, if the Media Termination Point Required check box is checked, Cisco Unified Communications Manager will allocate an MTP and the SIP trunk will negotiate DTMF according to RFC 2833. |
| Cisco TelePresence Conductor   | Cisco TelePresence Conductor provides intelligent conference administrative controls and is scalable, supporting device clustering for load balancing across MCUs and multiple device availability. Administrators can implement the Cisco TelePresence Conductor as either an appliance or a virtualized application on VMware with support for Cisco Unified Computing System (Cisco UCS) platforms or third-party-based platforms.  
Cisco TelePresence Conductor dynamically selects the most appropriate Cisco TelePresence resource for each new conference. Ad hoc, “MeetMe”, and scheduled voice and video conferences can dynamically grow and exceed the capacity of individual MCUs. Up to three Cisco TelePresence Conductor appliances or virtualized applications may be clustered to provide greater resilience. One Cisco TelePresence Conductor appliance or Cisco TelePresence Conductor cluster has a system capacity of 30 MCUs or 2400 MCU ports. |
Call Preservation

The call preservation feature of Cisco Unified Communications Manager ensures that an active call does not get interrupted when a Cisco Unified Communications Manager fails or when communication fails between the device and the Cisco Unified Communications Manager that set up the call.

Cisco Unified Communications Manager supports full call preservation for an extended set of Cisco Unified Communications devices. This support includes call preservation between Cisco Unified IP Phones, Media Gateway Control Protocol (MGCP) gateways that support Foreign Exchange Office (FXO) (non-loop-start trunks) and Foreign Exchange Station (FXS) interfaces, and, to a lesser extent, conference bridge, MTP, and transcoding resource devices.

Enable H.323 call preservation by setting the advanced service parameter, Allow Peer to Preserve H.323 Calls, to True.

The following devices and applications support call preservation. If both parties connect through one of the following devices, Cisco Unified Communications Manager maintains call preservation:

- Cisco Unified IP Phones
- SIP trunks
- Software conference bridge
- Software MTP
- Hardware conference bridge (Cisco Catalyst 6000 8 Port Voice E1/T1 and Services Module, Cisco Catalyst 4000 Access Gateway Module)
- Transcoder (Cisco Catalyst 6000 8 Port Voice E1/T1 and Services Module, Cisco Catalyst 4000 Access Gateway Module)
- Non-IOS MGCP gateways (Catalyst 6000 24 Port FXS Analog Interface Module, Cisco DT24+, Cisco DE30+, Cisco VG200)
- Cisco IOS H.323 gateways (such as Cisco 2800 series, Cisco 3800 series)
- Cisco IOS MGCP Gateways (Cisco VG200, Catalyst 4000 Access Gateway Module, Cisco 2620, Cisco 3620, Cisco 3640, Cisco 3660, Cisco 3810)
- Cisco VG248 Analog Phone Gateway

The following devices and applications do not support call preservation:

- Annunciator
- H.323 endpoints such as NetMeeting or third-party H.323 endpoints
- CTI applications
- TAPI applications
- JTAPI applications
## Call Preservation Scenarios

The below table lists and describes how call preservation is handled in various scenarios.

**Table 74: Call Preservation Scenarios**

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Call Preservation Handling</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified Communications Manager fails.</td>
<td>A Cisco Unified Communications Manager failure causes the call-processing function for all calls that were set up through the failed Cisco Unified Communications Manager to be lost. Cisco Unified Communications Manager maintains affected active calls until the end user hangs up or until the devices can determine that the media connection has been released. Users cannot invoke any call-processing features for calls that are maintained as a result of this failure.</td>
</tr>
<tr>
<td>Communication failure occurs between Cisco Unified Communications Manager and device.</td>
<td>When communication fails between a device and the Cisco Unified Communications Manager that controls it, the device recognizes the failure and maintains active connections. The Cisco Unified Communications Manager recognizes the communication failure and clears call-processing entities that are associated with calls in the device where communication was lost. The Cisco Unified Communications Managers still maintain control of the surviving devices that are associated with the affected calls. Cisco Unified Communications Manager maintains affected active calls until the end user hangs up or until the devices can determine that the media connection has been released. Users cannot invoke any call-processing features for calls that are maintained as a result of this failure.</td>
</tr>
</tbody>
</table>

**Note** The Phones or Devices will go into Call Preservation mode and Conference Bridge Profile gateway will have call entries until the user tears down the connection by placing the phones on-hook. Once the connectivity is back online, Conference Bridge Profile gateway will register to the CUCM, depending on **Switchback Guard time** configured under **sccp** settings on the gateway.
Call Preservation Handling

Scenario | Call Preservation Handling
---|---
Device failure (Phone, gateway, conference bridge, transcoder, MTP) | When a device fails, the connections that exist through the device stop streaming media. The active Cisco Unified Communications Manager recognizes the device failure and clears call-processing entities that are associated with calls in the failed device.

The Cisco Unified Communications Managers maintain control of the surviving devices that are associated with the affected calls. Cisco Unified Communications Manager maintains the active connections (calls) that are associated with the surviving devices until the surviving end users hang up or until the surviving devices can determine that the media connection has been released.

Conference Bridge Configuration Task Flow

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Conference Bridges, on page 479</td>
<td>Configure a hardware or software conference bridge to allow ad hoc and meet-me voice conferencing.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Service Parameters for Conference Bridges, on page 480</td>
<td>Perform this procedure when your network includes both Cisco IOS Conference Bridge and Cisco IOS Enhanced Conference Bridge.</td>
</tr>
</tbody>
</table>

Configure Conference Bridges

You must configure a hardware or software conference bridge to allow ad hoc and meet-me voice conferencing.

**Procedure**

1. From Cisco Unified CM Administration, choose **Media Resources > Conference Bridge**.
2. Click **Add New**.
3. Configure the fields in the **Conference Bridge Configuration** window. For detailed field descriptions, refer to the online help.
4. Click **Save**.
What to do next

If your network includes both Cisco IOS Conference Bridge and Cisco IOS Enhanced Conference Bridge, Configure Service Parameters for Conference Bridges, on page 480.

Configure Service Parameters for Conference Bridges

Perform this procedure when your network includes both Cisco IOS Conference Bridge and Cisco IOS Enhanced Conference Bridge.

Procedure

**Step 1** From Cisco Unified CM Administration, choose System > Service Parameters.
**Step 2** In the Service Parameter Configuration window, choose a server and choose the Cisco CallManager service.
**Step 3** In the Clusterwide Parameters (Features - Conference) section, set the following parameters to 6:
   - Maximum Ad Hoc Conference
   - Maximum MeetMe Conference Unicast

**Step 4** Click Save.
Flexible DSCP Marking and Video Promotion Overview

Devices and applications use Differentiated Services Code Point (DSCP) markings to indicate the Quality of Service (QoS) treatment of IP communications. For example, desktop video endpoints may use multimedia conferencing AF41 marking for video media streams, while high-definition video room systems may use real-time interactive CS4 marking. When an application sends and receives IP communications to and from the same type of application, the DSCP markings are symmetric, and the QoS treatments of the IP communications that each application sends and receives are the same. However, when an application sends and receives media and from a different type of application, the DSCP markings may be asymmetric, and the QoS treatments of the IP communications that each application sends and receives may be inconsistent. For example, the QoS treatment of the video media stream that a video room system receives from a desktop video endpoint may be inadequate to support the expected quality of the video room system.

Devices and applications are subjected to Call Admission Control (CAC) to ensure that adequate bandwidth is available for the duration of established sessions. The bandwidth that is utilized by established sessions is updated as the sessions begin and end. Attempts to establish new sessions that would exceed the available bandwidth are blocked. The amount of bandwidth available may be tracked independently for devices and applications of different types. For example, independent tracking of bandwidth may be available for desktop video endpoints and high-definition video room systems to send and receive video media streams.

When devices and applications of the same type send and receive communications, the same type of bandwidth deductions are made in each direction. However, when devices and applications of different types send and receive communications, different types of bandwidth deductions must be made in each direction. Moreover, the bandwidth deductions are usually symmetric in amount, by design, to reflect the usual behavior of an IP network. As a result, when devices and applications of different types send and receive communications, the total bandwidth deductions may be up to double the amount of network bandwidth that is actually utilized. This inconsistency in bandwidth accounting may cause attempts to establish new sessions to be blocked unnecessarily.
The Flexible DSCP Marking and Video Promotion feature allows you to configure a Video Promotion policy that reconciles the inconsistency in bandwidth accounting in favor of the application that receives more favorable CAC and QoS treatment. For example, if a session between a desktop video endpoint and a high-definition video room system is reconciled in favor of the video room system, then the reconciliation is deemed a promotion for the desktop video endpoint.

When reconciliation is in effect between devices and applications of different types, bandwidth is deducted only for the type of application that is favored by reconciliation. If sufficient bandwidth is available for a session of this type to be admitted, the device or application of the type that is not favored by reconciliation is instructed to change the DSCP markings that it uses to those that are used by the device or application of the type that is favored by reconciliation. For example, if a desktop video endpoint is promoted in a session with a high-definition video room system, bandwidth accounting takes place as if the desktop video endpoint were an application of the same type as the video room system. The desktop video endpoint is instructed to change its DSCP markings to those that are used by the video room system. The QoS treatment is consistent in both directions, bandwidth is deducted for a session between devices and applications of the same type as the video room system, and bandwidth is not deducted for a session between devices and applications of the same type as the desktop video endpoint.

When you activate the Flexible DSCP Marking and Video Promotion feature, Unified Communications Manager dynamically signals desktop video devices a Traffic Class Label that is indicative of the DSCP marking for each negotiated media stream.

Custom QoS Settings for Users

You can customize Quality of Service (QoS) settings within a SIP profile and apply those settings to your users. The SIP Profile Configuration window has been enhanced with the following types of QoS settings:

- Custom DSCP values for audio and video streams
- Custom UDP port ranges for audio and video streams

Custom DSCP Values for Audio and Video

You can configure DSCP values for audio and video calls within a SIP profile and apply them to the SIP phones that use that profile. The SIP Profile Configuration window includes custom DSCP settings for the following types of calls:

- Audio calls
- Video calls
- Audio portion of a video call
- TelePresence calls
- Audio portion of a TelePresence call

If your company has a set of employees, such as a sales force, or a CEO, who require higher QoS priority settings than the majority of your employees, you can use the SIP profile configurations to configure custom DSCP values for those users. The settings within the SIP profile override the corresponding clusterwide service parameter settings.
Custom UDP Port Ranges for Audio and Video

You can configure separate UDP port ranges for the audio stream and video stream of a SIP call. Because video typically requires considerably more bandwidth than audio, creating dedicated port ranges for each media type simplifies network bandwidth management. It also protects against audio stream degradation by guaranteeing that the audio stream will have a dedicated channel that is separate from the higher-bandwidth video stream.

You can apply this configuration by setting the Media Port Ranges field in the SIP profile to Separate Port Ranges for Audio and Video. You can then apply the configuration to a phone by associating the SIP profile to a phone.

Traffic Class Label

The Flexible DSCP and Video Promotion feature uses the Traffic Class Label (TCL) to instruct the SIP endpoint dynamically to mark its DSCP on a per call basis, based on the Video Promotion policy that you configure. Because TCL is a SIP Session Description Protocol (SDP) attribute that is defined per media line, the TCL and its associated DSCP markings can be different for the audio media line and the video media line of a video call. You can choose different DSCP markings for the audio stream and the video stream of the video call.

DSCP Settings Configuration Task Flow

Perform the following tasks to configure DSCP values and a video promotion policy for your network.

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Flexible DSCP Marking and Video Promotion Policy, on page 483</td>
<td>Configure a video promotion policy to handle the different types of video.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Custom QoS Policy for Users, on page 485</td>
<td>If your company has users that require higher priority than other users in your company, configure a SIP Profile that includes custom DSCP values for audio and video streams. For example, if your company has a telephone sales force or CEO whom require higher priority, you can apply the customized SIP profile to those users' phones.</td>
</tr>
</tbody>
</table>

Configure Flexible DSCP Marking and Video Promotion Policy

Follow these steps to configure a video promotion policy to handle the different types of video.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Service Parameters.
Step 2  From the Server drop-down list, choose the server where you want to configure the parameters.

Step 3  From the Service drop-down list, choose the Cisco CallManager (Active) service.

If the service does not display as active, ensure that the service is activated in Cisco Unified Serviceability.

Step 4  To configure a Video Promotion policy that promotes desktop video endpoints to immersive video endpoints, set the Use Video BandwidthPool for Immersive Video Calls parameter to False and set the Video Call QoS Marking Policy parameter to Promote to Immersive.

Step 5  To configure other parameters, scroll to the appropriate area of the Service Parameter Configuration window and update the parameter values. See Flexible DSCP Marking and Video Promotion Service Parameters, on page 484 for information about the service parameters and their configuration options.

Step 6  Click Save.

Flexible DSCP Marking and Video Promotion Service Parameters

For more information about the service parameters, click the parameter name or click the question mark (?) icon that displays in the Service Parameter Configuration window.

Note

Table 75: Flexible DSCP Marking and Video Promotion Service Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Clusterwide Parameters (System - QoS)</td>
<td>This section of service parameters includes clusterwide DSCP values for a wide range of audio and video call types, including DSCP for audio calls, video calls, the audio portion of a video call, TelePresence calls, and the audio portion of a TelePresence call. It is highly recommended that you keep these parameters set to the default value unless a Cisco support engineer instructs otherwise.</td>
</tr>
<tr>
<td>Clusterwide Parameters (Call Admission Control)</td>
<td></td>
</tr>
<tr>
<td>Video Call QoS Marking Policy</td>
<td>This parameter allows you to configure a Promote to Immersive policy that reconciles bandwidth allocation inconsistencies between a desktop video endpoint and a Cisco TelePresence immersive video endpoint in favor of the immersive endpoint. When promotion is performed, the audio and video bandwidth are reserved from the immersive bandwidth pool allocation. The policy of Promote to Immersive takes effect only for calls between an immersive video device and a desktop video device that supports flexible DSCP marking.</td>
</tr>
<tr>
<td>Clusterwide Parameters (System - Location and Region)</td>
<td></td>
</tr>
<tr>
<td>Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio)</td>
<td>This parameter specifies the default maximum total bit rate for each immersive video call within a particular region, when the Use System Default option is selected as the Max Immersive Video Call Bit Rate in the Region Configuration window for the relationship of the region with itself.</td>
</tr>
<tr>
<td>Parameter</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Default Interregion Max Immersive Video Call Bit Rate (Includes Audio)</td>
<td>This parameter specifies the default maximum total bit rate for each immersive video call between a particular region and another region, when the Use System Default option is selected as the Max Immersive Video Call Bit Rate in the Region Configuration window for the relationship of the region with the other region.</td>
</tr>
<tr>
<td>Use Video BandwidthPool for Immersive Video Calls</td>
<td>This parameter specifies whether Unified Communications Manager reserves bandwidth from the desktop video bandwidth pool for immersive video calls.</td>
</tr>
</tbody>
</table>

**Configure Custom QoS Policy for Users**

Perform the following tasks to set up a custom Quality of Service (QoS) policy for users. You may want to apply a custom policy if a set of users within your company has different QoS requirements from the rest of the company such as telephone sales force or a CEO.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Custom QoS Settings in SIP Profile, on page 485</td>
<td>Configure a SIP Profile with customized DSCP values and a UDP port range for audio and video streams.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Apply Custom QoS Policy to a Phone, on page 486</td>
<td>Apply the SIP Profile to a phone. The DSCP settings in the SIP Profile override the DSCP clusterwide service parameter settings.</td>
</tr>
</tbody>
</table>

**Configure Custom QoS Settings in SIP Profile**

Configure custom DSCP values and UDP port ranges for the phones that use this SIP Profile. You can use these settings to configure a customized QoS policy that you can apply to specific phones and users within your network. You may want to do this if you want to apply specific QoS settings to specific users within your enterprise, such as a sales force, or a CEO.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Device Settings > SIP Profile**.

**Step 2** Perform either of the following steps:
- Click **Find** and select an existing SIP Profile.
- Click **Add New** to create a new SIP Profile.

**Step 3** From the **Media Port Ranges** field, select whether you want to assign a single UDP port range that handles both audio and video media, or separate port ranges for audio and video streams.
- If you want to configure a single port range for audio and video media, enter the range of ports in the **Start Media Port** and **Stop Media Port** fields. The possible port values are between 2048 and 65535.
Apply Custom QoS Policy to a Phone

If you want separate port ranges for audio and video streams, enter the range of audio ports using the Start Audio Port and Stop Audio Port fields. Enter the range of video ports using the Start Video Port and Stop Video Port fields. The possible port values for each are between 2048 and 65535. The two port ranges must not overlap.

Step 4 In the following fields, configure customized DSCP values for audio and video streams.

- DSCP for Audio Calls
- DSCP for Video Calls
- DSCP for Audio Portion of Video Calls
- DSCP for TelePresence Calls
- DSCP for Audio Portion of TelePresence Calls

Note By default, each of the above fields is configured to use the value from a corresponding service parameter. If you assign new values, the new value overrides the service parameter setting.

Step 5 Complete the remaining fields in the SIP Profile Configuration window. For help with the fields and their settings, refer to the online help.

Step 6 Click Save.

Apply Custom QoS Policy to a Phone

Use this procedure to apply a SIP Profile that contains customized QoS settings, including DSCP values and a UDP port range for audio and video media. When you apply this SIP profile to a phone, the phone uses the custom settings from the SIP Profile.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.

Step 2 Perform any one of the following steps:
- Click Find and select an existing phone.
- Click Add New to create a new phone.

Step 3 From the SIP Profile drop-down list, select the SIP profile that you set up with the custom DSCP values and UDP port range values.

Step 4 Complete the remaining fields in the Phone Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 5 Click Save.

Flexible DSCP Marking and Video Promotion Interactions and Restrictions

You can perform:

- Flexible DSCP Marking and Video Promotion Interactions, on page 487
Flexible DSCP Marking and Video Promotion Interactions

Table 76: Flexible DSCP Marking and Video Promotion Interactions

<table>
<thead>
<tr>
<th>Device</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Intercluster Trunks</td>
<td>The Flexible DSCP Marking and Video Promotion feature is supported over SIP intercluster trunks.</td>
</tr>
<tr>
<td>Skinny Client Control Protocol (SCCP) Devices</td>
<td>The Flexible DSCP Marking and Video Promotion feature is supported for SCCP devices.</td>
</tr>
<tr>
<td>Pass-Through MTPs</td>
<td>If pass-through MTPs are inserted in a call, Unified Communications Manager signals the MTP to mark the packets with the DSCP marking that is expected from the endpoint device that originally emitted the packet for the video stream. If the two endpoints on a call use different DSCP markings (for example, a Cisco TelePresence immersive video endpoint and a desktop video endpoint without Video Promotion), the MTPs preserve the DSCP marking in each stream direction.</td>
</tr>
</tbody>
</table>

Flexible DSCP Marking and Video Promotion Restrictions

Table 77: Flexible DSCP Marking and Video Promotion Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunks and gateways</td>
<td>The Flexible DSCP Marking and Video Promotion feature is not supported over H.323 trunks and Media Gateway Control Protocol (MGCP) gateways.</td>
</tr>
<tr>
<td>Multilevel Precedence and Preemption</td>
<td>Cisco recommends that you do not use the Flexible DSCP Marking and Video Promotion feature with Multilevel Precedence and Preemption (MLPP) service calls. When you need MLPP service functionality, Cisco recommends that you set the Video Call QoS Marking Policy and Use Video BandwidthPool for Immersive Video Calls service parameters to their default values. With default values for the Video Call QoS Marking Policy and Use Video BandwidthPool for Immersive Video Calls service parameters, Unified Communications Manager and endpoints use MLPP DSCP markings for the media packets.</td>
</tr>
<tr>
<td>SIP video endpoints</td>
<td>The Flexible DSCP Marking and Video Promotion feature is dependent on desktop SIP video endpoint support. Currently, only Cisco DX650 series SIP phones provide the required endpoint support.</td>
</tr>
</tbody>
</table>
Flexible DSCP Marking and Video Promotion Restrictions
CHAPTER 66

Configure Transcoders and Media Termination Points

- Transcoders and Media Termination Points Overview, on page 489
- Transcoders and MTPs Configuration Task Flow, on page 494
- Transcoders and MTPs Interactions and Restrictions, on page 498

Transcoders and Media Termination Points Overview

Transcoders

A transcoder is a device that performs codec conversion, converting an input stream from one codec into an output stream that uses a different codec. For example, a transcoder can take a G.711 stream and convert it to a G.729 stream in real time. When the two endpoints in a call use different voice codecs, Cisco Unified Communications Manager invokes a transcoder into the media path. The transcoder converts the data streams between the two incompatible codecs so that communication can occur. The transcoder is invisible to either the user or the endpoints that are involved in a call.

Transcoder resources is managed by the Media Resource Manager (MRM).

Note

The transcoder supports transcoding between G.711 and all codecs, including G.711, when functioning as a transcoder and when providing MTP/TRP functionality.

Transcoders with MTP Functionality

In addition to codec conversion, a transcoder can provide the same functionality as a media termination point (MTP). In cases where both transcoder functionality and MTP functionality are needed, the system allocates a transcoder due to the fact that transcoders can provide both sets of functionality simultaneously. If only MTP functionality is required, the system allocates either a transcoder or an MTP from the resource pool. The choice of resource will be determined by the media resource groups.

If a software MTP resource is not available when it is needed, the call tries to connect without using an MTP resource and MTP/TRP services. If hardware transcoder functionality is required (to convert one codec to another) and a transcoder is not available, the call will fail.
Transcoders and the Media Resource Manager

All Cisco Unified Communications Manager nodes can access transcoders through the Media Resource Manager (MRM). The MRM manages access to transcoders.

The MRM makes use of Cisco Unified Communications Manager media resource groups and media resource group lists. The media resource group list allows transcoders to communicate with other devices in the assigned media resource group, which in turn, provides management of resources within a cluster.

A transcoding control process gets created for each transcoding device that is defined in the database. The MRM keeps track of the transcoding resources and advertises their availability throughout the cluster.

Transcoders as Media Termination Points

Hardware-based transcoding resources also support Media Termination Point (MTP) and/or Trust Relay Point (TRP) functionality. In this capacity, when Cisco Unified Communications Manager determines that an endpoint in a call requires an MTP or TRP, it can allocate a transcoding resource and insert it into the call, where it acts like an MTP transcoding.

Cisco Unified Communications Manager supports MTP and TRP and transcoding functionality simultaneously. For example, if a call originates from a Cisco Unified IP Phone (located in the G723 region) to NetMeeting (located in the G711 region), one transcoding resource supports MTP and transcoding functionality simultaneously.

If a software MTP resource is not available when it is needed, the call tries to connect without using an MTP resource and MTP/TRP services. If hardware transcoding functionality is required (to convert one codec to another) and a transcoding is not available, the call will fail.

Note

The transcoding supports transcoding between G.711 and all codecs, including G.711, when functioning as a transcoding and when providing MTP/TRP functionality.

Transcoder Types

Transcoder types in Cisco Unified Communications Manager Administration are listed in the following table.

Note

The transcoding supports transcoding between G.711 and all codecs, including G.711, when functioning as a transcoding and when providing MTP/TRP functionality.
### Table 78: Transcoder Types

<table>
<thead>
<tr>
<th>Transcoder Type</th>
<th>Description</th>
</tr>
</thead>
</table>
| Cisco Media Termination Point Hardware                | This type, which supports the Cisco Catalyst 4000 WS-X4604-GWY and the Cisco Catalyst 6000 WS-6608-T1 or WS-6608-E1, provides the following number of transcoding sessions:  
  For the Cisco Catalyst 4000 WS-X4604-GWY  
  • For transcoding to G.711-16 MTP transcoding sessions  
  For the Cisco Catalyst 6000 WS-6608-T1 or WS-6608-E1  
  • For transcoding from G.723 to G.711/For transcoding from G.729 to G.711-24 MTP transcoding sessions per physical port; 192 sessions per module |
| Cisco IOS Media Termination Point (hardware)          | This type, which supports the Cisco 2600XM, Cisco 2691, Cisco 3725, Cisco 3745, Cisco 3660, Cisco 3640, Cisco 3620, Cisco 2600, and Cisco VG200 gateways, provides the following number of transcoding sessions:  
  Per NM-HDV  
  • Transcoding from G.711 to G.729-60  
  • Transcoding from G.711 to GSM FR/GSM EFR- 45 |
| Cisco IOS Enhanced Media Termination Point (hardware) | Per NM-HD  
  This type, which supports Cisco 2600XM, Cisco 2691, Cisco 3660, Cisco 3725, Cisco 3745, and Cisco 3660 Access Routers, provides the following number of transcoding sessions:  
  • Transcoding for G.711 to G.729a/G.729ab/GSMFR-24  
  • Transcoding for G.711 to G.729/G.729b/GSM EFR-18  
  Per NM-HDV2  
  This type, which supports Cisco 2600XM, Cisco 2691, Cisco 3725, Cisco 3745, and Cisco 3660 Access Routers, provides the following number of transcoding sessions:  
  • Transcoding for G.711 to G.729a/G.729ab/GSMFR-128  
  • Transcoding for G.711 to G.729/G.729b/GSM EFR-96 |
Transcoder Type | Description
---|---
Cisco Media Termination Point (WS-SVC-CMM) | This type provides 64 transcoding sessions per daughter card that is populated: 64 transcoding sessions with one daughter card, 128 transcoding sessions with two daughter cards, 192 transcoding sessions with three daughter cards, and 256 transcoding sessions with four daughter cards (maximum).

This type provides transcoding between any combination of the following codecs:
• G.711 a-law and G.711 mu-law
• G.729 annex A and annex B
• G.723.1
• GSM (FR)
• GSM (EFR)

Transcoder Failover and Fallback

The following items describe the transcoder device recovery methods when the transcoder is registered to a Cisco Unified Communications Manager node that goes inactive:

• If the primary Cisco Unified Communications Manager node fails, the transcoder attempts to register with the next available node in the Cisco Unified Communications Manager Group that is specified for the device pool to which the transcoder belongs.

• The transcoder device reregisters with the primary Cisco Unified Communications Manager node as soon as it becomes available.

• A transcoder device unregisters with a Cisco Unified Communications Manager node that becomes unreachable. The calls that were on that node will register with the next Cisco Unified Communications Manager node in the list.

• If a transcoder attempts to register with a new Cisco Unified Communications Manager node and the register acknowledgment is never received, the transcoder registers with the next node in the list.

Transcoder devices will unregister and then disconnect after a hard or soft reset. After the reset completes, the devices reregister with the primary Cisco Unified Communications Manager node.

Media Termination Points

Media Termination Points (MTP) allow Unified Communications Manager to relay calls that are routed through SIP or H.323 endpoints or gateways. Media Termination Points extend supplementary services, such as call hold, call transfer, call park, and conferencing, that are normally not available when a call is routed to an H.323 endpoint. For H.323 supplementary services, MTPs are only required for endpoints that do not support EmptyCapability Set (ECS) or FastStart. All Cisco and other third party other endpoints that support ECS and FastStart do not require an MTP.

An MTP device always registers with its primary Unified Communications Manager if that Unified Communications Manager is available and informs the Unified Communications Manager about the number of MTP resources it supports. You can register multiple MTPs with the same Unified Communications Manager. When more than one MTP is registered with a Unified Communications Manager, that Cisco Unified Communications Manager controls the set of resources for each MTP.
For example, consider MTP server 1 as configured for 48 MTP resources, and the MTP server 2 as configured for 24 resources. If both MTPs register with the same Unified Communications Manager, that Unified Communications Manager maintains both sets of resources for a total of 72 registered MTP resources.

When the Unified Communications Manager determines that a call endpoint requires an MTP, it allocates an MTP resource from the MTP that has the least active streams. That MTP resource gets inserted into the call on behalf of the endpoint. MTP resource use remains invisible to both the users of the system and to the endpoint on whose behalf it was inserted. If an MTP resource is not available when it is needed, the call connects without using an MTP resource, and that call does not have supplementary services.

**MTP Failover and Fallback**

This section describes how MTP devices failover and fallback when the Cisco Unified Communications Manager to which they are registered becomes unreachable:

- If the primary Cisco Unified Communications Manager fails, the MTP attempts to register with the next available Cisco Unified Communications Manager in the Cisco Unified Communications Manager Group that is specified for the device pool to which the MTP belongs.
- The MTP device reregisters with the primary Cisco Unified Communications Manager as soon as it becomes available after a failure and is currently not in use.
- The system maintains the calls or conferences that were active in call preservation mode until all parties disconnect. The system does not make supplementary services available.
- If an MTP attempts to register with a new Cisco Unified Communications Manager and the register acknowledgment is never received, the MTP registers with the next Cisco Unified Communications Manager.

The MTP devices unregister and then disconnect after a hard or soft reset. After the reset completes, the devices reregister with the Cisco Unified Communications Manager.

**Software Media Termination Point Type**

Software Media Termination Point type in Cisco Unified Communications Manager Administration is listed in the following table.

<table>
<thead>
<tr>
<th>Software MTP Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Media Termination Point Software</td>
<td>A single MTP provides a default of 48 MTP (user configurable) resources, depending on the speed of the network and the network interface card (NIC). For example, a 100-MB Network/NIC card can support 48 MTP resources, while a 10-MB NIC card cannot. For a 10-MB Network/NIC card, approximately 24 MTP resources can be provided; however, the exact number of MTP resources that are available depends on the resources that other applications on that PC are consuming, the speed of the processor, network loading, and various other factors.</td>
</tr>
</tbody>
</table>
Transcoders and MTPs Configuration Task Flow

### Configure Transcoders

#### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Determine the number of transcoder resources that are needed and the number of transcoder devices that are needed to provide these resources.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Transcoders, on page 495</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Add Transcoder to Media Resource Group, on page 495</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Restart the transcoder devices.</td>
</tr>
</tbody>
</table>

#### Configure Transcoders, on page 494

**Purpose**

To configure a transcoder, follow this step. Transcoders convert an input stream from one codec into an output stream that uses a different codec.

**Command or Action**

- Configure Transcoders, on page 495
- Add Transcoder to Media Resource Group, on page 495

**Purpose**

If you need to configure a software MTP, follow this step. Software MTPs allow Cisco Unified Communications Manager to relay calls that are routed through SIP or H.323 endpoints or gateways.

**Command or Action**

- Configure a Software MTP, on page 496
- Configure Media Termination Points, on page 497
- Add Software MTP to Media Resource Group, on page 497

**Purpose**

If you need to configure a software MTP, follow this step. Software MTPs allow Cisco Unified Communications Manager to relay calls that are routed through SIP or H.323 endpoints or gateways.

**Command or Action**

- Configure Transcoders, on page 495

**Purpose**

Configure transcoders to convert an input stream from one codec into an output stream that uses a different codec.

**Command or Action**

- Add Transcoder to Media Resource Group, on page 495

**Purpose**

Add the new transcoders to the appropriate media resource groups.

**Command or Action**

- Restart the transcoder devices.

See the transcoder documentation for details.
Configure Transcoders

A transcoder is a device that converts an input stream from one codec into an output stream that uses a different codec.

Before you begin

The Cisco IP Voice Media Streaming service must be running for the IVR to be active.

Determine the number of transcoder resources that are needed and the number of transcoder devices that are needed to provide these resources.

Procedure

Step 1 Log into Cisco Unified CM Administration and choose Media Resources > Transcoder.
Step 2 Do either of the following:
   • Click Find and select an existing transcoder.
   • Click Add New.
Step 3 Select the Transcoder Type.
Step 4 Enter the MAC Address of the transcoder.
Step 5 Assign a Device Pool from the drop-down menu.
Step 6 Check the Trusted Relay Point check box if you want to make this transcoder available as a trusted relay point.
Step 7 Click Save.

Add Transcoder to Media Resource Group

Before you begin

Configure Transcoders, on page 495

Procedure

Step 1 Choose Media Resources > Media Resource Group.
Step 2 Click Find to display the list of configured Media Resource Groups.
Step 3 Click on the required Media Resource Group. The Media Resource Group Configuration window displays.
Step 4 Select the transcoder from the list of available media resources and add it to the Selected Media Resources list.
Step 5 Click Save.
Step 6 Navigate to Media Resources > Media Resource Group.
Step 7 In the Find and List Transcoders window, check the check boxes next to the transcoders that you want to synchronize. To choose all transcoders in the window, check the check box in the matching records title bar.
Step 8 Click Apply Config to Selected.
Synchronize Transcoder

The Apply Configuration Information dialog box displays.

**Step 9**

Click **OK**.

---

**What to do next**

Restart the transcoder device.

---

**Synchronize Transcoder**

To synchronize a transcoder with the most recent configuration changes, perform the following procedure, which applies any outstanding configuration settings in the least-intrusive manner possible. (For example, a reset/restart may not be required on some affected devices.).

**Procedure**

**Step 1**

Choose **Media Resources** > **Transcoder**.

The Find and List Transcoders window displays.

**Step 2**

Choose the search criteria to use.

**Step 3**

Click **Find**.

The window displays a list of transcoders that match the search criteria.

**Step 4**

Check the check boxes next to the transcoders that you want to synchronize. To choose all transcoders in the window, check the check box in the matching records title bar.

**Step 5**

Click **Apply Config to Selected**.

The Apply Configuration Information dialog box displays.

**Step 6**

Click **OK**.

---

**Configure a Software MTP**

This procedure describes the steps to configure a software MTP. For information about configuring a hardware MTP, see **Configure Transcoders, on page 494**.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Media Termination Points, on page 497</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Add Software MTP to Media Resource Group, on page 497</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Restart the media termination point devices.</td>
</tr>
</tbody>
</table>
Configure Media Termination Points

Use this procedure to configure a software Media Termination Point (MTP).

**Before you begin**

The Cisco IP Voice Media Streaming service must be running for the software Media Termination Point (MTP) to be active.

Determine the number of MTP resources that are needed and the number of MTP devices that are needed to provide these resources.

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose Media Resources > Media Termination Point.

**Step 2**  
Do either of the following:
- Click Find and select an existing MTP.
- Click Add New to create a new MTP.

**Step 3**  
Assign a Media Termination Point Name.

**Step 4**  
Assign a Device Pool.

**Step 5**  
Check the Trusted Relay Point check box if you want to designate this MTP as a Trusted Relay Point (TRP).

**Step 6**  
Click Save.

Add Software MTP to Media Resource Group

**Before you begin**

Configure Media Termination Points, on page 497

**Procedure**

**Step 1**  
Choose Media Resources > Media Resource Group.

**Step 2**  
Click Find to display the list of configured Media Resource Groups.

**Step 3**  
Click on the required Media Resource Group. The Media Resource Group Configuration window displays.

**Step 4**  
Select the transcoder from the list of Available Media Resources and add it to the Selected Media Resources list.

**Step 5**  
Click Save.

**What to do next**

Restart the media termination point device.
# Transcoders and MTPs Interactions and Restrictions

## Transcoder Interactions and Restrictions

### Transcoder Deletion
You cannot delete a transcoder that is assigned to a media resource group. To find out which media resource groups are using the transcoder, click **Dependency Records** from the **Related Links** drop-down list box on the **Transcoder Configuration** window and click **Go**. The Dependency Records Summary window displays information about media resource groups that are using the transcoder. To find out more information about the media resource group, click the media resource group, and the Dependency Records Details window displays. If the dependency records are not enabled for the system, the dependency records summary window displays a message. If you try to delete a transcoder that is in use, Cisco Unified Communications Manager displays a message. Before deleting a transcoder that is currently in use, you must remove the transcoder from the media resource group(s) to which it is assigned.

### Failover and Fallback
Transcoder failover and fallback works as follows:

- If the primary Cisco Unified Communications Manager node fails, the transcoder attempts to register with the next available node in the Cisco Unified Communications Manager Group that is specified for the device pool to which the transcoder belongs.

- The transcoder device rerегистers with the primary Cisco Unified Communications Manager node as soon as it becomes available.

- A transcoder device unregisters with a Cisco Unified Communications Manager node that becomes unreachable. The calls that were on that node will register with the next Cisco Unified Communications Manager node in the list.

- If a transcoder attempts to register with a new Cisco Unified Communications Manager node and the register acknowledgment is never received, the transcoder registers with the next node in the list.

Transcoder devices will unregister and then disconnect after a hard or soft reset. After the reset completes, the devices reregister with the primary Cisco Unified Communications Manager node.

<table>
<thead>
<tr>
<th>Interactions or Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transcoder Deletion</td>
<td>You cannot delete a transcoder that is assigned to a media resource group. To find out which media resource groups are using the transcoder, click <strong>Dependency Records</strong> from the <strong>Related Links</strong> drop-down list box on the <strong>Transcoder Configuration</strong> window and click <strong>Go</strong>. The Dependency Records Summary window displays information about media resource groups that are using the transcoder. To find out more information about the media resource group, click the media resource group, and the Dependency Records Details window displays. If the dependency records are not enabled for the system, the dependency records summary window displays a message. If you try to delete a transcoder that is in use, Cisco Unified Communications Manager displays a message. Before deleting a transcoder that is currently in use, you must remove the transcoder from the media resource group(s) to which it is assigned.</td>
</tr>
<tr>
<td>Failover and Fallback</td>
<td>Transcoder failover and fallback works as follows:</td>
</tr>
<tr>
<td></td>
<td>- If the primary Cisco Unified Communications Manager node fails, the transcoder attempts to register with the next available node in the Cisco Unified Communications Manager Group that is specified for the device pool to which the transcoder belongs.</td>
</tr>
<tr>
<td></td>
<td>- The transcoder device rerегистers with the primary Cisco Unified Communications Manager node as soon as it becomes available.</td>
</tr>
<tr>
<td></td>
<td>- A transcoder device unregisters with a Cisco Unified Communications Manager node that becomes unreachable. The calls that were on that node will register with the next Cisco Unified Communications Manager node in the list.</td>
</tr>
<tr>
<td></td>
<td>- If a transcoder attempts to register with a new Cisco Unified Communications Manager node and the register acknowledgment is never received, the transcoder registers with the next node in the list.</td>
</tr>
</tbody>
</table>

Transcoder devices will unregister and then disconnect after a hard or soft reset. After the reset completes, the devices reregister with the primary Cisco Unified Communications Manager node.
## Media Termination Points Interactions and Restrictions

### Table 79: Media Termination Points Interactions and Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IP Voice Streaming Application</td>
<td>You can activate only one Cisco IP Voice Streaming Application per server. To provide more MTP resources, you can activate the Cisco IP Voice Streaming application on additional networked servers. Cisco strongly recommends that you do not activate the Cisco IP Voice Streaming Media Application on a Cisco Unified Communications Manager with a high call-processing load because it can adversely affect the performance of the Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Registering with Cisco Unified Communications Manager</td>
<td>Each MTP can register with only one Cisco Unified Communications Manager at a time. The system may have multiple MTPs, each of which may be registered to one Cisco Unified Communications Manager, depending on how your system is configured.</td>
</tr>
</tbody>
</table>
| Failover and Fallback                           | This section describes how MTP devices failover and fallback when the Cisco Unified Communications Manager to which they are registered becomes unreachable:  

  * If the primary Cisco Unified Communications Manager fails, the MTP attempts to register with the next available Cisco Unified Communications Manager in the Cisco Unified Communications Manager Group that is specified for the device pool to which the MTP belongs.  
  * The MTP device reregisters with the primary Cisco Unified Communications Manager as soon as it becomes available after a failure and is currently not in use.  
  * The system maintains the calls or conferences that were active in call preservation mode until all parties disconnect. The system does not make supplementary services available.  
  * If an MTP attempts to register with a new Cisco Unified Communications Manager and the register acknowledgment is never received, the MTP registers with the next Cisco Unified Communications Manager.  

The MTP devices unregister and then disconnect after a hard or soft reset. After the reset completes, the devices reregister with the Cisco Unified Communications Manager. |
PART IX

Register Devices

- Register Devices Overview, on page 503
- Configure TFTP Servers, on page 505
- Update Device Defaults, on page 513
- Configure Autoregistration, on page 517
- Manual Phone Registration, on page 525
- Configure Self-Provisioning, on page 529
CHAPTER 67

Register Devices Overview

• About Registering Devices, on page 503
• Registering Devices, on page 503

About Registering Devices

The chapters in this section describe the tasks that you perform to register new endpoint devices and to set up proxy TFTP servers for your endpoints and gateway devices.

You can choose to register new phones manually or use autoregistration. To register more than 100 phones, use the Bulk Administration Tool (BAT). For more information, see Cisco Unified Communications Manager Bulk Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

Note

You cannot create new settings using BAT, but you can configure phone parameters when you use the BAT to register phones. Make sure that phone settings such as device pool, location, calling search space, button template, and softkey templates have already been configured using Cisco Unified Communications Manager Administration.

Registering Devices

Complete the following task flows to register devices for your system.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 TFTP Server Configuration Task Flow, on page 508</td>
<td>Configure the proxy Trivial File Transfer Protocol (TFTP) server that provides the configuration files for endpoints in your network.</td>
</tr>
</tbody>
</table>
### Registering Devices

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>(Optional) Update Device Defaults Task Flow, on page 513</td>
<td>Modify the device load, device pool, and the phone button template values that are applied to endpoints when they register.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Autoregistration Task Flow, on page 517</td>
<td>Enable autoregistration for your network. Because of the inherent security risk of allowing devices to register automatically on the network, we recommend that you disable autoregistration as soon as you have registered your new endpoints.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Manual Device Registration Task Flow, on page 525</td>
<td>Manually register an IP phone and assign a new directory number.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Self-Provisioning Configuration Task Flow, on page 530</td>
<td>Optional. If you want your end users to be able to provision their own phones without the aid of an administrator, configure self-provisioning.</td>
</tr>
</tbody>
</table>
Configure TFTP Servers

- Proxy TFTP Deployment Overview, on page 505
- TFTP Server Configuration Task Flow, on page 508

Proxy TFTP Deployment Overview

Use a proxy Trivial File Transfer Protocol (TFTP) server to provide the configuration files that endpoints in your network need, such as: dial plans, ringer files, and device configuration files. A TFTP server can be installed in any cluster in your deployment and can service requests from endpoints on multiple clusters. The DHCP scope specifies the IP address of the proxy TFTP server to use to get the configuration files.

Redundant and Peer Proxy TFTP Servers

In a single cluster deployment, the cluster must have at least one proxy TFTP server. You can add another proxy TFTP server to the cluster for redundancy. The second proxy TFTP server is added in option 150 for IPv4. For IPv6, you add the second proxy TFTP server to TFTP Server Addresses sub-option type 1 in the DHCP scope.

In a multiple cluster deployment, you can specify up to three remote proxy TFTP servers as peer clusters of the primary proxy TFTP server. This is useful if you want to configure only one proxy TFTP server for many DHCP scopes, or have only one DHCP scope. The primary proxy TFTP server provides the configuration files for all phones and devices in the network.

You must create a peer relationship between each remote proxy TFTP server and the primary proxy TFTP server.

Tip

When you configure peer relationships between the remote proxy TFTP servers in your network, keep the relationships hierarchical. Ensure that the peer proxy TFTP servers on the remote clusters do not point to each other to avoid possible looping. For example, if the primary node A has a peer relationship with nodes B and C. You should not create a peer relationship between nodes B and C. If you do, then you have created a loop.

Proxy TFTP

In multi-cluster systems, the proxy TFTP service is able provide TFTP files from multiple clusters via a single primary TFTP server. The proxy TFTP can serve as a single TFTP reference for scenarios where a single
subnet or VLAN contains phones from multiple clusters or in any scenario where multiple clusters share the same DHCP TFTP option (150).

The Proxy TFTP service functions as a single-level hierarchy is as illustrated. More complicated multi-level hierarchies are not supported.

*Figure 4: Proxy TFTP Single-Level Hierarchy*

In the above illustration, a group of devices contacts the Primary TFTP server for their configuration files. When it receives a request for TFTP from a device, the primary TFTP looks into its own local cache for the configuration file as well as any other remotely configured clusters such as Remote Cluster A, B, C, or N (any other remote clusters configured).

It is possible to configure any number of remote clusters on the primary TFTP server; however, each remote cluster may contain only up to 3 TFTP IP addresses. The recommended design for redundancy is 2 TFTP servers per cluster, and thus 2 IP addresses per remote cluster on the Primary TFTP server for redundancy.

**TFTP Support for IPv4 and IPv6 Devices**

We recommend that you enable IPv4 phones and gateways to use the DHCP custom option 150 to discover the TFTP server IP address. Using option 150, gateways and phones discover the TFTP server IP address. For more information, see the documentation that came with your device.

In an IPv6 network, we recommend that you use the Cisco vendor-specific DHCPv6 information to pass the TFTP server IPv6 address to the endpoint. With this method, you configure the TFTP server IP address as the option value.

If you have some endpoints that use IPv4 and some that use IPv6, we recommend that you use DHCP custom option 150 for IPv4 and use the TFTP Server Addresses sub-option type 1, a Cisco vendor-specific information option, for IPv6. If the endpoint obtains an IPv6 address and sends a request to the TFTP server while the TFTP server is using IPv4 to process requests, the TFTP server does not receive the request because the TFTP
server is not listening for the request on the IPv6 stack. In this case, the endpoint cannot register with Cisco Unified Communications Manager.

There are alternative methods that you could use for your IPv4 and IPv6 devices to discover the IP address of the TFTP server. For example, you could use DHCP option 066 or CiscoCM1 for your IPv4 devices. For your IPv6 devices, other methods include using TFTP Service sub-option type 2 or configuring the IP address of the TFTP server on the endpoint. These alternative methods are not recommended. Consult your Cisco service provider before using any alternative methods.

Endpoints and Configuration Files for TFTP Deployments

SCCP phones, SIP phones and gateways, request a configuration file when they initialize. An updated configuration file gets sent to the endpoint whenever you change the device configuration.

The configuration file contains information such as a prioritized list of Unified Communications Manager nodes, the TCP ports used to connect to those nodes, as well other executables. For some endpoints, the configuration file also contains locale information and URLs for phone buttons, such as: messages, directories, services, and information. Configuration files for gateways contain all the configuration information that the device requires.

Security Considerations for Proxy TFTP

Cisco Proxy TFTP servers handle both signed and unsigned requests and run in either nonsecure mode or mixed mode. The Proxy TFTP Server searches the local file system or database when a phone requests for a file and if not found, sends a request to remote clusters. When the phone requests the server for a common file with names such as ringlist.xml.sgn, locale file, and so on, the server sends a local copy of the file instead of the file itself from the home cluster of the phone.

When receiving files from Proxy TFTP, the phone rejects the file due to a signature verification failure because the file has the signature of the proxy server which doesn't match the Initial Trust List (ITL) of the phone. To resolve this issue, you can either disable Security By Default (SBD) for the Phone or perform the bulk certificate export procedure between the Proxy and remote clusters so that the Trust Verification Service (TVS) can verify the certificate for the Phone.

To disable Security by Default, see "Update ITL File for Cisco Unified IP Phones" section the Security Guide for Cisco Unified Communications Manager.

Proxy TFTP in Mixed Mode

TFTP servers on remote clusters that are running in mixed mode must have the primary Proxy TFTP server certificates added to their Cisco Certificate Trust List (CTL) file. Otherwise, endpoints that are registered to a cluster where security is enabled will be unable to download the files that they need. To achieve this update CTL file after performing bulk import-export of certificates.

For more information, see "Bulk Certificate Export" section in the Security Guide for Cisco Unified Communications Manager when migrating IP phones between clusters to perform the bulk certificate export.

Moving Phones Between Clusters in Proxy TFTP Environment

When moving phones from one Remote Cluster to another in a Proxy TFTP environment, perform the following:

1. Add Phone details to Remote Cluster B (destination cluster).
2. Delete Phone details from Remote Cluster A (source cluster).
The phone’s configuration in the Proxy TFTP takes 30 minutes to expire. To avoid any file not found response, you can restart Proxy Cluster’s TFTP services.

3. Reset Phones to download configuration files from Remote Cluster B and register to Remote Cluster B.

TFTP Server Configuration Task Flow

You can let the system dynamically configure the proxy TFTP server if you have Extension Mobility Cross Cluster (EMCC) configured for your cluster. If you don’t, then you can set up the TFTP server and set the security mode manually.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Set up the TFTP server using one of the following methods:</td>
</tr>
<tr>
<td></td>
<td>• Configure TFTP Server Dynamically, on page 509</td>
</tr>
<tr>
<td></td>
<td>• Configure TFTP Server Manually, on page 509</td>
</tr>
<tr>
<td>If you have Intercluster Lookup Service (ILS) configured, you can set up your TFTP server dynamically.</td>
<td></td>
</tr>
<tr>
<td>If you don’t have EMCC configured, set up your TFTP server manually. You must indicate if the cluster is secured or non-secured. The cluster is treated as non-secure by default.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>(Optional) Update the CTL File for TFTP Servers, on page 510</td>
</tr>
<tr>
<td>Install the CTL client plug-in and add the primary proxy TFTP server to the Cisco Certificate Trust List (CTL) file of all proxy TFTP servers in all remote clusters that are operating in mixed-mode.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>(Optional) See the documentation that supports your endpoint device.</td>
</tr>
<tr>
<td>Add the proxy TFTP servers to the Trust Verification List (TVL) of all remote endpoints if your proxy TFTP deployment has remote clusters.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>(Optional) Modify Non-Configuration Files for the TFTP Server, on page 511</td>
</tr>
<tr>
<td>You can modify non-configuration files that the endpoints request from the proxy TFTP server.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>(Optional) Stop and Start the TFTP service, on page 511</td>
</tr>
<tr>
<td>Stop and restart the TFTP service on the proxy TFTP node if you have uploaded modified non-configuration files for your endpoints.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>(Optional) See the documentation that supports your DHCP server.</td>
</tr>
<tr>
<td>For multiple cluster deployments, modify the DHCP scope for individual remote nodes to include the IP address of the primary proxy TFTP server.</td>
<td></td>
</tr>
</tbody>
</table>
Configure TFTP Server Dynamically

You can configure a Cisco proxy TFTP server dynamically if you have Intercluster Lookup Service (ILS) configured for your network.

Before you begin


Procedure

In Cisco Unified Communications Manager Administration, choose Advanced Features > Cluster View > Update Remote Cluster Now. The TFTP server is automatically configured for the cluster.

What to do next

You must add any remote proxy TFTP servers to the Trust Verification Lists (TVL) of the endpoints; otherwise, they will not accept the configuration files from the proxy TFTP server that is on a remote cluster. See the documentation that supports your endpoint device for instructions.

Configure TFTP Server Manually

To configure TFTP in your network when you don't have EMCC configured, you must use the manual procedure.

You set up peer relationships between the primary proxy TFTP server and other TFTP servers from the Cluster View. You can add up to three peer TFTP servers.

Each remote TFTP server in the proxy TFTP deployment must include a peer relationship to the primary proxy TFTP server. To avoid creating a loop, ensure that the peer TFTP servers on the remote clusters do not point to each other.

Procedure

Step 1

Create a remote cluster. Perform the following actions:

a) From Cisco Unified CM Administration, select Advanced Features > Cluster View.

b) Click Add New. The Remote Cluster Configuration window appears.

c) Enter a cluster ID and a Fully Qualified Domain Name (FQDN) of up to 50 characters for the TFTP server, then click Save.

   Valid values for the cluster ID include alphanumerical characters, period (.), and hyphen (-). Valid values for the FQDN include alphanumerical characters, period (.), dash (-), asterisk (*), and space.

d) (Optional) In the Remote Cluster Service Configuration window, enter a description of up to 128 characters for the remote cluster.
Step 2  Check the TFTP check box to enable TFTP for the remote cluster.
Step 3  Click TFTP.
Step 4  In the Remote Cluster Service Manually Override Configuration window, select Manually configure remote service addresses.
Step 5  Enter the IP addresses of the TFTP server to create a peer relationships to those TFTP servers.
        You can enter up to three TFTP server IP addresses.
Step 6  (Optional) Check the Cluster is Secure check box if the proxy TFTP server is deployed in a secured cluster.
Step 7  Click Save.

What to do next
You must add any remote TFTP servers to the Trust Verification Lists (TVL) of the endpoints; otherwise, they will not accept the configuration files from the proxy TFTP server that is on a remote cluster. See the documentation that supports your endpoint device for instructions.

Update the CTL File for TFTP Servers
Update the CTL file from publisher node by running `utils ctl` in each cluster which is in mixed mode. Make sure that a complete security network is attained between the Proxy TFTP server and all the clusters, that is bulk import and export exchange of certificates between Proxy and remote clusters.

While using CTLClient, you must add the primary TFTP server or the IP address of the primary TFTP server to the Cisco Certificate Trust List (CTL) file of all TFTP servers in remote clusters that are running in mixed mode. This is necessary so that endpoints in security-enabled clusters can successfully download their configuration files.

For more information about security and using the Cisco CTL CLI, see the "About Cisco CTL Setup" section in the Security Guide for Cisco Unified Communications Manager.

Procedure

Step 1  From Cisco Unified CM Administration, choose Application > Plugins.
Step 2  Click Find to list of all the plug-ins that you can install.
Step 3  Click the Download link for the Cisco CTL Client.
        The system installs the client that digitally signs certificates stored on the TFTP server.
Step 4  Reboot the TFTP server.
Modify Non-Configuration Files for the TFTP Server

You can modify a non-configuration file, such as a load file or RingList.xml, that the endpoints request from the proxy TFTP server. After you complete this procedure, upload the modified files to the TFTP directory of the proxy TFTP server.

Procedure

**Step 1** In Cisco Unified Communications Operating System Administration, select Software Upgrades > TFTP File Management. The TFTP File Management window appears.

**Step 2** Click Upload File. The Upload File pop-up appears.

**Step 3** Perform one of the following actions:
- Click Browse to browse to the directory location of the file to upload.
- Paste the full directory path of the updated file in to the Directory field.

**Step 4** Click Upload File or click Close to exit without uploading the file.

What to do next

Stop and restart the Cisco TFTP service on the proxy TFTP node using Cisco Unified Serviceability Administration.

Stop and Start the TFTP service

Use the following procedure to stop and restart the TFTP service on the proxy TFTP node.


Procedure

**Step 1** In Cisco Unified Serviceability, select Tools > Control Center - Feature Services.

**Step 2** In the Control Center–Feature Services window, select the proxy TFTP node in the Server drop-down list.

**Step 3** Select the TFTP service in the CM Services area and click Stop. The status changes to reflect the updated status.

**Tip** To see the latest status of services, click Refresh.

**Step 4** Select the TFTP service in the CM Services area, then click Start. The status changes to reflect the updated status.
Stop and Start the TFTP service
Update Device Defaults

- Device Defaults Overview, on page 513
- Update Device Defaults Task Flow, on page 513

Device Defaults Overview

Each device that registers with a Cisco Unified Communications Manager node is configured with the defaults for that type of device. Device defaults are applied to all auto-registering devices in the cluster. After registration, you can change the device's configuration.

You cannot create new device defaults or delete existing ones, but you can change the default settings that get applied to devices that auto-register.

These are the device default settings that you can change.

- Device load
- Device pool
- Phone button template

Installing a Cisco Unified Communications Manager automatically sets device defaults.

Update Device Defaults Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Update Device Default Settings, on page 514</td>
<td>You can change the default settings that are applied to devices that auto-register with a Cisco Unified Communications Manager node. Each type of device has a specific set of defaults.</td>
</tr>
</tbody>
</table>
Update Device Default Settings

Use this procedure to configure Device Default settings that allow you to assign default firmware loads, default device pools, softkey templates and the registration method: auto registration or activation codes.

Before you begin

Before updating the device default settings, perform any of the following tasks that apply to your system.

- Add new firmware files for the devices to the TFTP server.
- If you use device defaults to assign a firmware load that does not exist in the directory, those devices will fail to load the assigned firmware.
- Configure new device pools. If the device is a phone, configure new phone templates.

Procedure

Step 1
In Cisco Unified CM Administration, select Device > Device Settings > Device Defaults.

Step 2
In the Device Defaults Configuration window, modify the applicable settings for the type of device that you want to update, then click Save. For field descriptions, see the online help.

- Load Information
- Device Pool
- Phone Template

Step 3
Click the Reset icon that appears to the left of the device name to reset all the devices of that type and load the new defaults to all devices of that type on all nodes in the cluster.

If you do not reset all devices, then only new devices that auto-register on the node are configured with the updated default values.

Device Defaults Settings

Table 80: Device Defaults Settings

<table>
<thead>
<tr>
<th>Field Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Type</td>
<td>This field displays the type of device to which the defaults apply.</td>
</tr>
<tr>
<td>Protocol</td>
<td>This field displays the protocol that is used for this type of device.</td>
</tr>
<tr>
<td>Load Information</td>
<td>Enter the ID number of the firmware load that is used with a particular type of hardware device. If you install an upgrade or patch load, you must update the load information for each type of device that uses the new load.</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Choose the device pool to associate with each type of device. The device pool defines common characteristics for all devices in the pool.</td>
</tr>
<tr>
<td>Field Name</td>
<td>Description</td>
</tr>
<tr>
<td>----------------</td>
<td>------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Phone Template</td>
<td>Choose the phone button template that each type of Cisco IP Phone uses. The template defines the function of the keys on the phone.</td>
</tr>
</tbody>
</table>
Configure Autoregistration

- Autoregistration Overview, on page 517
- Configure Autoregistration Task Flow, on page 517

Autoregistration Overview

Autoregistration allows Unified Communications Manager to automatically assign directory numbers to new phones when you plug those phones in to your network.

Autoregistration is enabled on secure mode now. This enhancement provides greater security for your system because you can secure your cluster while provisioning new phones. It also simplifies the registration process because you don't have to disable cluster security to register new phones.

If you create a device pool that allows only 911 (emergency) and 0 (operator) calls, you can use that to prevent unauthorized endpoints from connecting to your network when autoregistration is enabled. New endpoints can register to this pool, but their access is limited. Unauthorized access by rogue devices that continuously boot in and attempt to register to your network is prevented. You can move a phone that has auto-registered to a new location and assign it to a different device pool without affecting its directory number.

The system doesn't know whether the new phones that are auto-registering are running SIP or SCCP, so you must specify this when you enable autoregistration. Devices that support both SIP and SCCP (such as Cisco IP Phones 7911, 7940, 7941, 7960, 7961, 7970, and 7971) auto-register with the protocol that is specified in the enterprise parameter called Auto Registration Phone Protocol.

Devices that support only a single protocol will auto-register with that protocol. The Auto Registration Phone Protocol setting is ignored. For example, any Cisco IP Phones that support SCCP only will auto-register with SCCP even if the Auto Registration Phone Protocol parameter is set to SIP.

We recommend that you use autoregistration to add fewer than 100 phones to your network. To add more than 100 phones, use the Bulk Administration Tool (BAT). For more information, see Cisco Unified Communications Manager Bulk Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

Configure Autoregistration Task Flow

Enabling autoregistration carries a security risk. Enable autoregistration only for brief periods while you add new endpoints to the network.
## Configure a Partition for Autoregistration

Configure a route partition to use specifically for autoregistration to limit auto-registered phones to internal calls only.

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure a Partition for Autoregistration, on page 518</td>
<td>Configure a route partition to use specifically for autoregistration to limit auto-registered phones to internal calls only.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure a Calling Search Space for Autoregistration, on page 519</td>
<td>Configure a calling search space to use specifically for autoregistration to limit auto-registered phones to internal calls only.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Device Pool for Autoregistration, on page 520</td>
<td>Create a device pool that uses the calling search space that is configured for autoregistration.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Set the Device Protocol Type for Autoregistration, on page 521</td>
<td>Use this procedure to set the protocol to SCCP or SIP to match the type of phones you are auto-registering.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Enable Autoregistration, on page 521</td>
<td>Enable autoregistration on the node to use for autoregistration and set the <strong>Auto-registration Cisco Unified Communications Manager Group</strong> parameter to enable autoregistration for the Cisco Unified Communications Manager group that is to be used for autoregistration.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Disable Autoregistration, on page 523</td>
<td>Disable autoregistration for the node as soon as you are finished registering new devices.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Reuse Autoregistration Numbers, on page 524</td>
<td>Optional. Autoregistration numbers for devices that have been disabled can be reused. When you reset the range of autoregistration directory numbers, you force the system to search again from the starting number. Available directory numbers are reused.</td>
</tr>
</tbody>
</table>

## Configure a Partition for Autoregistration

Configure a route partition to use specifically for autoregistration to limit auto-registered phones to internal calls only.

### Procedure

1. From Cisco Unified CM Administration, choose **Call Routing > Class of Control > Partition**.
2. Click **Add New** to create a new partition.
3. In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan. Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.
4. Enter a comma (,) after the partition name and enter a description of the partition on the same line.
Configure a Calling Search Space for Autoregistration

Configure a calling search space to use specifically for autoregistration to limit auto-registered phones to internal calls only.

Before you begin
Configure a Partition for Autoregistration, on page 518

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Calling Search Space.
Step 2 Click Add New.
Step 3 In the Name field, enter a name. Ensure that each calling search space name is unique to the system. The name can include up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).
Step 4 In the Description field, enter a description. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).
Step 5 From the Available Partitions drop-down list, perform one of the following steps:
  • For a single partition, select that partition.
Configure a Device Pool for Autoregistration

You can use the Default device pool for autoregistration or configure separate device pools for SIP and SCCP devices to use for autoregistration.

To configure the Default device pool for autoregistration, assign the Default Cisco Unified Communications Manager Group and the autoregistration calling search space (CSS) to the Default device pool. If you choose to configure a separate default device pool for SIP and SCCP devices, use the default device pool values.

Before you begin
Configure a Calling Search Space for Autoregistration, on page 519

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose System > Device Pool.
Step 2 To modify the Default device pool for autoregistration, perform the following actions:
   a) Click Find, then select Default from the list of device pools.
   b) In the Device Pool Configuration window, select the CSS to be used for autoregistration in the Calling Search Space for Auto-registration field, then click Save.
Step 3 To create a new device pool for autoregistration, perform the following actions:
   a) Click Add New.
   b) In the Device Pool Configuration window, enter a unique name for the device pool.
      You can enter up to 50 characters, which include alphanumeric characters, periods (.), hyphens (-), underscores (_), and blank spaces.
   c) Set the following fields to match the Default device pool. See the online help for field descriptions.
      • In Cisco Unified Communications Manager Group, select Default.
      • In Date/Time Group, select CMLocal
      • In Region, select Default.
d) Select the CSS to be used for autoregistration in the Calling Search Space for Auto-registration field, then click Save.

What to do next
Set the Device Protocol Type for Autoregistration, on page 521

Set the Device Protocol Type for Autoregistration

If you have SIP and SCCP devices to auto-register, you must first set the Auto Registration Phone Protocol parameter to SCCP and install all the devices that are running SCCP. Then change the Auto Registration Phone Protocol parameter to SIP and auto-register all the devices that are running SIP.

Before you begin
Configure a Device Pool for Autoregistration, on page 520

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>In Cisco Unified Communications Manager Administration, select System &gt; Enterprise Parameters.</td>
</tr>
<tr>
<td>2</td>
<td>In the Enterprise Parameters Configuration window, select either SCCP or SIP in the Auto Registration Phone Protocol drop-down list, then click Save.</td>
</tr>
</tbody>
</table>

What to do next
Enable Autoregistration, on page 521

Enable Autoregistration

When you enable autoregistration, you must specify a range of directory numbers that get assigned to the new endpoints as they connect to the network. As each new endpoint connects, the next available directory number is assigned. After all the available autoregistration directory numbers are used up, no more endpoints can auto-register.

New endpoints auto-register with the first Unified Communications Manager node in the group that has the Auto-Registration Cisco Unified Communications Manager Group setting enabled. That node then automatically assigns each auto-registered endpoint to a default device pool according to the device type.

Before you begin
Set the Device Protocol Type for Autoregistration, on page 521

- Create a device pool, calling search space, and route partition that restricts the access of devices that are auto-registering to allow only internal calls.
- Ensure that directory numbers are available in the autoregistration range.
- Ensure that there are enough license points available to register the new phones.
Procedure

Step 1  From Cisco Unified Communications Manager Administration, select System > Cisco Unified CM, then click Find in the Find and List Cisco Unified Communications Managers window.

Step 2  Select the Cisco Unified Communications Manager in the cluster to use for autoregistration.

Step 3  In the Cisco Unified CM Configuration window, configure the autoregistration parameters for the node in the Auto-registration Information section, then click Save. For more information on the fields and their configuration options, see the system Online Help.

   a) Select the universal device template to use for autoregistration from the drop-down list.

      If no universal device template is created for autoregistration, you can select Default Universal Device Template. Make sure that the selected template specifies the device pool that is to be used for autoregistration from User Management > User/Phone Add > Universal Device Template.

   b) Select the universal line template to use for autoregistration from the drop-down list.

      If no universal line template is created for autoregistration, you can select Default Universal Line Template. Make sure that the selected template specifies the calling search space and the route partition that are to be used for autoregistration from User Management > User/Phone Add > Universal Line Template.

   c) Enter the starting and ending directory numbers into the Starting Directory Number and Ending Directory Number fields.

      Setting the starting and ending directory numbers to the same value disables autoregistration.

   d) Uncheck Auto-registration Disabled on this Cisco Unified Communications Manager to enable autoregistration for this node.

      Always enable or disable autoregistration on only the selected Unified Communications Manager node. If you switch the autoregistration function to another node in the cluster, you must reconfigure the Unified Communications Manager nodes, the Default Unified Communications Manager group, and the default device pools that you used.

Step 4  Select System > Cisco Unified CM Group, then click Find in the Find and List Cisco Unified Communications Manager Groups window.

Step 5  Select the Unified Communications Manager group to enable for autoregistration.

      In most cases, the name of this group is Default. You can choose a different Cisco Unified Communications Manager group. The group must have at least one node selected.

Step 6  In the Cisco Unified CM Group Configuration window for that group, select Auto-registration Cisco Unified Communications Manager Group to enable autoregistration for the group, then click Save.
Ensure that the Selected Cisco Unified Communications Managers list contains the node that you configured for autoregistration. Use the arrows to move the node to appear in the list. The Unified Communications Manager nodes get selected in the order in which they are listed. Save your changes.

**Step 7**  
Install the devices that you want to auto-register.

---

**Note**  
You can proceed to reconfigure the auto-registered phones and assign them to their permanent device pools. The directory number that is assigned to the phone does not change when you change the phone location.

**Note**  
To register phones of a different type, change the device protocol type and install those devices before disabling autoregistration.

---

**Disable Autoregistration**

Disable autoregistration for the node as soon as you are finished registering new devices.

**Before you begin**

Enable Autoregistration, on page 521

**Procedure**

**Step 1**  
In Cisco Unified Communications Manager Administration, select System > Cisco Unified CM, then click Find in the Find and List Cisco Unified CM window.

**Step 2**  
Select the Cisco Unified Communications Manager from the list of nodes.

**Step 3**  
In the Cisco Unified CM Configuration widow for the selected node, check the Auto-registration Disabled on this Cisco Unified Communications Manager check box to disable autoregistration for this node, then click Save.

**Tip**  
Setting the same value in the Starting Directory Number and Ending Directory Number fields also disables autoregistration.

**What to do next**

Optional. If you manually changed the directory number of an auto-registered device, or if you delete that device from the database, you can reuse the directory number. For details, see Reuse Autoregistration Numbers, on page 524.
Reuse Autoregistration Numbers

When you connect a new device to the network, the system assigns the next available autoregistration directory number to that device. If you manually change the directory number of an auto-registered device, or if you delete that device from the database, the autoregistration directory number of that device can be reused.

When a device attempts to auto-register, the system searches the range of autoregistration numbers that you specified and tries to find the next available directory number to assign to the device. It begins the search with the next directory number in sequence after the last one that was assigned. If it reaches the ending directory number in the range, the system continues to search from the starting directory number in the range.

You can reset the range of autoregistration directory numbers and force the system to search from the starting number in the range.

Procedure

Step 1  In Cisco Unified Communications Manager Administration, select System > Cisco Unified Communications Manager

Step 2  Select the Cisco Unified Communications Manager to reset for autoregistration.

Step 3  Write down the current settings in the Starting Directory Number and Ending Directory Number fields.

Step 4  Click Auto-registration Disabled on this Cisco Unified Communications Manager, then click Save.

New phones cannot auto-register while autoregistration is disabled.

Step 5  Set the Starting Directory Number and Ending Directory Number fields to their previous values, then click Save.

Tip  You could set the fields to new values.
Manual Phone Registration

- Manual Phone Registration Overview, on page 525
- Manual Device Registration Task Flow, on page 525

Manual Phone Registration Overview

To manually register a new Cisco IP Phone, you must add the phone to the Unified Communications Manager node using Cisco Unified Communications Manager Administration, then configure the directory number for the phone.

You should have already set up the new phone with the proxy TFTP server IP address so that the new phone knows how to locate the Unified Communications Manager node. See the Cisco IP Phone Administration Guide for your phone series.

Related Topics
- Proxy TFTP Deployment Overview, on page 505

Manual Device Registration Task Flow

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>Command or Action</strong></td>
</tr>
<tr>
<td></td>
<td>See the Cisco IP Phone Administration Guide for your phone series</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Add a Phone to the System Manually, on page 526</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure a Directory Number Manually for a Phone, on page 526</td>
</tr>
</tbody>
</table>
Add a Phone to the System Manually

Manually add a new phone to the Cisco Unified Communications Manager node.

**Procedure**

**Step 1**
In Cisco Unified Communications Manager Administration, select Device > Phone, then click Add New.

**Step 2**
In the Add a New Phone window, select your phone model in the Phone Type field, then click Next.

**Step 3**
In the Phone Configuration window, select the protocol type for your device in the Select the device protocol field, then click Next.

**Step 4**
In the Device Information area, perform the following actions.
   a) Enter a name in the Device Name field.
      The name entered here must match the Device Name that is configured on your phone. See the documentation that supports your endpoint device for more information.
   b) Select a device pool for the phone from the list of device pools.
   c) Select the phone button template to use from the list of phone button templates.

**Step 5**
In the Protocol Specific Information area, select the non-secure profile for your type of phone in the Device Security Profile field.

**Step 6**
Click Save.

**What to do next**
Configure a Directory Number Manually for a Phone, on page 526

Configure a Directory Number Manually for a Phone

There are multiple ways to manually add and configure a directory number (DN) using Cisco Unified Communications Manager Administration.

- From the Directory Number Configuration window using Call Routing > Directory Number.
- From the Phone Configuration window using Device > Phone when you select either Line [1] - Add a new DN or Line [2] - Add a new DN link in the Association Information area.
- From the Phone Configuration window using Call Routing > Phone after you add the phone under call routing.
- From the CTI Route Point Configuration window when you configure a CTI route point using Device > CTI Route Point.

This procedure assumes that you are configuring a DN for a new phone using the Phone Configuration window that appeared after you added the new phone to the Unified Communications Manager node.

Only the settings that apply to your phone model display using this method.
You can configure phone features at the same time that you add the new DN for the phone. To see all available DN settings, you must access the Directory Number Configuration window from call routing in the user interface.

Before you begin

The phone is added to the node. The Phone Configuration window should still be visible for the new phone that you are registering.

If your system uses partitions, collect the route partition and calling search space information to use for the new phone.

Procedure

Step 1
Click Line [1] - Add a new DN in the Association area of the Phone Configuration window.

Tip
If the Phone Configuration window is not already visible, select Device > Phone, then click Find and select the phone from the list of phones.

Step 2
In the Directory Number Configuration window, enter a dialable phone number in the Directory Number field.

Step 3
(Optional) Select a partition in the Route Partition field.

Step 4
(Optional) Select a calling search space in the Calling Search Space field in the Directory Number Settings area.

Step 5
(Optional) Configure other directory number features as applicable for the new phone, then click Save.

For example, if you already know the user name for the new phone, you can enter that in the Display (Caller ID) field. See the online help for field descriptions.

Related Topics

Class of Service, on page 115
Configure Self-Provisioning

Self-Provisioning Overview

The Self-Provisioning feature helps you provision phones for your network by giving end users the ability to provision their own phones without contacting an administrator. If the system is configured for self-provisioning, and an individual end user is enabled for self-provisioning, then end user can provision a new phone by plugging the phone into the network and follow the specified few prompts. Cisco Unified Communications Manager configures the phone and the phone line by applying pre-configured templates.

Self-provisioning can be used either by administrators to provision phones on behalf of their end users, or end users can use self-provisioning to provision their own phones.

Self-provisioning is supported whether the cluster security setting is nonsecure or mixed mode.

Security Modes

You can configure self-provisioning in one of two modes:

- Secure mode—In secure mode, users or administrators must be authenticated in order to access self-provisioning. End users can be authenticated against their password or PIN. Administrators can enter a pre-configured authentication code.

- Non-secure mode—In non-secure mode, users or administrators can enter their userID, or a self-provisioning ID, in order to associate the phone to a user account. Non-secure mode is not recommended for day-to-day use.

Configuration through Universal Line and Device Templates

Self-provisioning uses the universal line template and universal device template configurations to configure provisioned phones and phone lines for an end user. When a user provisions their own phone, the system references the user profile for that user and applies the associated universal line template to the provisioned phone line and the universal device template to the provisioned phone.
### Self-Provisioning Phones

When the feature is configured, you can provision a phone by doing the following:

- Plug the phone into the network.
- Dial the self-provisioning IVR extension.
- Follow the prompts to configure the phone, and associate the phone to an end user. Depending on how you have configured self-provisioning, the end user may need to enter the user password, PIN, or an administrative authentication code.

#### Tip

If you are provisioning a large number of phones on behalf of your end users, configure a speed dial on the universal device template that forwards to the self-provisioning IVR extension.

### Self-Provisioning Prerequisites

Before your end users can use self-provisioning, your end users be configured with the following items:

- Your end users must have a primary extension.
- Your end users must be associated to a user profile or feature group template that includes a universal line template, universal device template. The user profile must be enabled for self-provisioning.

### Self-Provisioning Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>Activate Services for Self-Provisioning, on page 531</strong></td>
<td>In Cisco Unified Serviceability, activate the <strong>Self-Provisioning IVR and CTI Manager</strong> services.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>Enable Autoregistration for Self-Provisioning, on page 531</strong></td>
<td>Enable autoregistration parameter for self-provisioning</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>Configure CTI Route Point, on page 531</strong></td>
<td>Configure a CTI route point to handle the self-provisioning IVR service.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td><strong>Assign a Directory Number to the CTI Route Point, on page 532</strong></td>
<td>Configure the extension that users will dial in order to access the self-provisioning IVR and associate that extension to the CTI route point.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>Configure Application User for Self-Provisioning, on page 532</strong></td>
<td>Configure an application user for the self-provisioning IVR. Associate the CTI route point to the application user.</td>
</tr>
</tbody>
</table>
Activate Services for Self-Provisioning

Use this procedure to activate the services that support the Self-Provisioning feature. Ensure that both the Self-Provisioning IVR and Cisco CTI Manager services are running.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 6</td>
<td>Configure the System for Self-Provisioning, on page 533</td>
<td>Configure self-provisioning settings for your system, including associating the application user and CTI route point to the self-provisioning IVR.</td>
</tr>
</tbody>
</table>

Enable Autoregistration for Self-Provisioning

Use this procedure for self-provisioning, you must configure the auto-registration parameters on the publisher.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified Serviceability, choose Tools &gt; Service Activation.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list, select the publisher node and click Go.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Under CM Services, check Cisco CTI Manager.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Under CTI Services, check Self Provisioning IVR.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
<td></td>
</tr>
</tbody>
</table>

Configure CTI Route Point

Use this procedure to configure a CTI Route Point for the Self-Provisioning IVR.
Assign a Directory Number to the CTI Route Point

Use this procedure to set up the extension that users will dial in to access the self-provisioning IVR. You must associate this extension to the CTI route point that you want to use for self-provisioning.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > CTI Route Points.
Step 2  Click Find and select the CTI route point that you set up for self-provisioning.
Step 4  In the Directory Number field, enter the extension that you want users to dial to access the Self-Provisioning IVR service.
Step 5  Click Save.
Step 6  Complete the remaining fields in the Directory Number Configuration window. For more information with the fields and their settings, see the online help.
Step 7  Click Save.

Configure Application User for Self-Provisioning

You must set up an application user for the self-provisioning IVR and associate the CTI route point that you created to the application user.
Procedure

Step 1
From Cisco Unified CM Administration, choose User > Application User.

Step 2
Perform either of the following steps:
   a) To select an existing application user, click Find and select the application user.
   b) To create a new application user, click Add New.

Step 3
In the User ID text box, enter a unique ID for the application user.

Step 4
Select a BLF Presence Group for the application user.

Step 5
Associate the CTI route point that you created to the application user by performing the following steps:
   a) If the CTI route point that you created does not appear in the Available Devices list box, click Find More Route Points.
      The CTI route point that you created displays as an available device.
   b) In the Available Devices list, select the CTI route point that you created for self-provisioning and click the down arrow.
      The CTI route point displays in the Controlled Devices list.

Step 6
Complete the remaining fields in the Application User Configuration window. For help with the fields and their settings, see the online help.

Step 7
Click Save.

Configure the System for Self-Provisioning

Use this procedure to configure your system for self-provisioning. Self-provisioning provides users in your network with the ability to add their own desk phone through an IVR system, without contacting an administrator.

Note
In order to use the self-provisioning feature, your end users must also have the feature enabled in their user profiles.

Procedure

Step 1
From Cisco Unified CM Administration, choose User Management > Self-Provisioning.

Step 2
Configure whether you want the self-provisioning IVR to authenticate end users by clicking one of the following radio buttons:
   • Require Authentication—In order to use the self-provisioning IVR, end users must enter their password, PIN, or a system authentication code.
   • No Authentication Required—End users can access the self-provisioning IVR without authenticating.

Step 3
If the self-provisioning IVR is configured to require authentication, click one of the following radio buttons to configure the method whereby the IVR authenticates end users:
   • Allow authentication for end users only—End users must enter their password or PIN.
• **Allow authentication for users (via Password/PIN) and Administrators (via Authentication Code)—**End Users must enter an authentication code. If you choose this option, configure the authentication code by entering an integer between 0 and 20 digits in the Authentication Code text box.

**Step 4** In the IVR Settings list boxes, use the arrows to select the Language that you prefer to use for IVR prompts. The list of available languages depends on the language packs that you have installed on your system. Refer to the Downloads section of cisco.com if you want to download additional language packs.

**Step 5** From the CTI Route Points drop-down list, choose the CTI route point that you have configured for your self-provisioning IVR.

**Step 6** From the Application User drop-down list, choose the application user that you have configured for self-provisioning.

**Step 7** Click Save.
PART X

Configure Advanced Call Handling

- Advanced Call Handling Overview, on page 537
- Configure QoS with APIC-EM Controller, on page 541
- Configure Call Control Discovery, on page 547
- Configure External Call Control, on page 557
- Configure Call Queuing, on page 567
- Configure Call Throttling, on page 579
- Calling Party Normalization, on page 583
- Configure Logical Partitioning, on page 593
- Configure Geolocation and Location Conveyance, on page 603
- Configure Location Awareness, on page 609
- Configure Automated Alternate Routing, on page 617
- Configure AS-SIP Endpoints, on page 619
- Configure Multilevel Precedence and Preemption, on page 631
Advanced Call Handling Overview

- About Advanced Call Handling, on page 537
- Advanced Call Handling Configuration, on page 537

About Advanced Call Handling

The chapters in this part describe different ways to configure advanced call handling in your system. With the functions outlined in this part, you can configure how your system handles a call at any point in the call flow at a more granular level than basic call handling features such as call forwarding. The task flow in this part lists each call handling function, describes the purpose for configuring it, and links to the applicable chapter that provides further details.

Advanced Call Handling Configuration

Complete the following task flows to configure advanced call handling for your system.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>APIC-EM Controller Configuration Task Flow, on page 542</td>
<td>In order to manage network quality of service (QoS) for SIP calls, deploy a Cisco Application Policy Infrastructure controller Enterprise module (APIC-EM). APIC-EM applies DSCP markings to media flows created by communication sessions among Cisco Unified Communications Manager-managed SIP endpoints and trunks. Applying DSCP markings to media flows ensures that audio and video media will not be blocked by other lower priority network traffic such as email, print jobs, and software downloads.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Call Control Discovery Configuration Task Flow, on page 547</td>
<td>Configure call control discovery to advertise the Cisco Unified Communications Manager to other call control entities that use the Service</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td>Advertisement Framework (SAF) network. These call control entities can use the advertised information to dynamically configure their routing operations for the call.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td><strong>External Call Control Configuration Task Flow, on page 558</strong></td>
<td></td>
</tr>
<tr>
<td>Configure external call control to have an adjunct route server make call-routing decisions for your system. Unified Communications Manager issues a route request to an adjunct route server, which provides instructions about how to route the call, along with any additional call treatment to apply.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td><strong>Call Queuing Task Flow, on page 568</strong></td>
<td></td>
</tr>
<tr>
<td>Configure call queueing to place callers in a queue until hunt members are available to answer them.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td><strong>Call Throttling Configuration, on page 580</strong></td>
<td></td>
</tr>
<tr>
<td>Configure call throttling to automatically throttle or deny new call attempts when system conditions can cause users to experience a delay in the interval between going off hook and receiving a dial tone. We recommend that you not modify call throttling parameters unless advised to do so by Cisco customer support.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td><strong>Calling Party Normalization Configuration Task Flow, on page 585</strong></td>
<td></td>
</tr>
<tr>
<td>Configure calling party normalization to reformat incoming phone numbers so that they display on the recipient's phone as globalized or localized phone numbers. Use this feature to improve callback functionality when a call is routed to multiple geographic locations, and to map a global calling party number to its localized variant so that a phone can return a call without modifying the directory number in the call log directories on the phone.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 7</td>
<td><strong>Logical Partitioning Configuration Task Flow, on page 593</strong></td>
<td></td>
</tr>
<tr>
<td>Configure logical partitioning to satisfy regulatory requirements in markets where toll bypass is forbidden. For example, you can configure a policy prevent users from initiating restricted calls by using midcall features such as conference join and redirect.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 8</td>
<td><strong>Geolocation and Location Conveyance Task Flow, on page 603</strong></td>
<td></td>
</tr>
<tr>
<td>Specify a geolocation for every device and communicate geolocation information across clusters. Geolocations assign a civic address to devices so that communication between devices can be controlled based on legal requirements in certain countries.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Step</td>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------</td>
<td>------------------------------------------------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>9</td>
<td>Location Awareness Configuration Task Flow, on page 611</td>
<td>Location Awareness allows administrators to determine the physical location from which a phone connects to the company network.</td>
</tr>
<tr>
<td>10</td>
<td>AAR Configuration Task Flow, on page 617</td>
<td>Configure your system to automatically reroute calls through the PSTN or other networks when your system blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.</td>
</tr>
<tr>
<td>11</td>
<td>Multilevel Precendence and Preemption Task Flow, on page 631</td>
<td>Configure Multilevel Precedence and Preemption (MLPP) if you want to allow validated users to place priority calls. If necessary, these users can preempt lower priority phone calls.</td>
</tr>
</tbody>
</table>
CHAPTER 74

Configure QoS with APIC-EM Controller

- APIC-EM Controller Overview, on page 541
- APIC-EM Controller Prerequisites, on page 542
- APIC-EM Controller Configuration Task Flow, on page 542

APIC-EM Controller Overview

The APIC-EM Controller provides a centralized system for managing network traffic so that you always have the bandwidth to maintain communications, even in congested networks. You can configure Cisco Unified Communications Manager to use the APIC-EM Controller to manage SIP media flows thereby providing the following benefits:

- Centralizes QoS management, thereby eliminating the need for endpoints to assign DSCP values.
- Applies differential QoS treatment for different media flows. For example, you can prioritize audio over video to ensure that basic audio communication is always maintained, even when network bandwidth is low.
- External QoS setting in the SIP Profile allows you to target which users will use the APIC-EM. For example, you may have Cisco Jabber users use the APIC-EM to manage media flows, while Cisco Unified IP Phone users use the DSCP settings in Cisco Unified Communications Manager.

SIP Media Flow Management

For SIP calls that use APIC-EM, Cisco Unified Communications Manager sends the policy request to the APIC-EM Controller at the call outset notifying the APIC-EM of the media flow that is being set up. The policy request contains information about the call, including the IP address and ports for source and destination devices, the media type for the flow and the protocol.

The APIC-EM notifies the switch at the beginning of the call flow of the DSCP values for the associated media flows. The switch inserts those DSCP values into individual media packets, overwriting any values that the endpoint inserts. If a gateway in the call flow experiences congestion, that gateway sends through the packets with the higher DSCP values first. This ensures that high priority audio and video streams are not blocked by lower-priority network traffic such as email, print jobs, or software downloads. When the call ends, Cisco Unified Communications Manager notifies the APIC-EM and the APIC-EM notifies the switch to delete the flow.
External QoS Support

In order for Cisco Unified Communications Manager to use the APIC-EM to manage media flows, the External QoS parameter must be enabled at both the system level, via a clusterwide service parameter, and at the device level, via the SIP Profile.

APIC-EM Controller Prerequisites

Before using APIC-EM, you must do the following:

- Configure DSCP priority for different SIP media flows in Cisco Unified Communications Manager. For details, see DSCP Settings Configuration Task Flow, on page 483.
- Configure the APIC-EM controller hardware within your network. For details, see the hardware documentation that comes with the APIC-EM controller.

APIC-EM Controller Configuration Task Flow

Complete these tasks on Cisco Unified Communications Manager to enable APIC-EM Controller to manage SIP media flows.

Before you begin

- Review APIC-EM Controller Prerequisites, on page 542.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
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</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Upload APIC-EM Controller Certificate, on page 543</td>
<td>Upload the APIC-EM certificate into Cisco Unified OS Administration.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure HTTPS Connection to APIC-EM Controller, on page 544</td>
<td>Configure an HTTP Profile that points to the APIC-EM service.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Enable External QoS Service for System, on page 544</td>
<td>Enable the External QoS Enable service parameter to configure the system to use the APIC-EM to manage media flows. The service parameter must be enabled for devices to use the APIC-EM for SIP media flow management. Note You must also enable external QoS within the SIP Profile for devices that will use the APIC-EM for SIP media flow management.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure External QoS Service at SIP Profile Level, on page 545</td>
<td>Enable external QoS within a SIP Profile. All devices that use this SIP Profile will be able to use the APIC-EM to manage SIP media flows</td>
</tr>
</tbody>
</table>
### Configure Advanced Call Handling

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 5</strong> Assign SIP Profile to Phones, on page 546</td>
<td>You can use the SIP Profile setting to configure which devices and device types you want the APIC-EM to manage media flows.</td>
</tr>
<tr>
<td></td>
<td>Associate the external QoS-enabled SIP Profile to a phone.</td>
</tr>
</tbody>
</table>

## Configure the APIC-EM Controller

Use this procedure on the APIC-EM Controller to add Cisco Unified Communications Manager as a user. APIC-EM's role-based access control feature provides Cisco Unified Communications Manager with access to APIC-EM resources.

### Procedure

**Step 1**
On the APIC-EM Controller, choose **Settings > Internal Users**.

**Step 2**
Create a new user with the following role: **ROLE_POLICY_ADMIN**. Keep track of the username and password that you enter because you must enter identical credentials in Cisco Unified Communications Manager's **HTTP Profile** window.

**Step 3**
Go to the **Discovery** tab and add a discovery with CDP or the IP address range of the available devices.

**Step 4**
Select the **Device Inventory** tab and select the reachable devices.

**Step 5**
Click on **Set Policy Tag**.

**Step 6**
Create a policy tag and set it for the devices.

**Step 7**
On the **EasyQoS** tab, select the policy that you created and enable **DynamicQoS**.

### What to do next

Upload APIC-EM Controller Certificate, on page 543

## Upload APIC-EM Controller Certificate

Use this procedure to upload the APIC-EM controller certificate into Cisco Unified Communications Manager.

### Procedure

**Step 1**
From Cisco Unified OS Administration, choose **Security > Certificate Management**.

**Step 2**
Click **Upload Certificate/Certificate Chain**. The **Upload Certificate/Certificate Chain** popup window appears.

**Step 3**
From the **Certificate Purpose** drop-down list, choose **CallManager-trust**.

**Step 4**
Enter a **Description** for the certificate.

**Step 5**
Click **Browse** to search for, and select, the certificate.
Configure HTTPS Connection to APIC-EM Controller

Use this procedure to set up an HTTP Profile to connect Cisco Unified Communications Manager to the APIC-EM Controller. In this connection, Cisco Unified Communications Manager acts as an HTTP user and the APIC-EM acts as the HTTP server.

**Before you begin**

Upload APIC-EM Controller Certificate, on page 543

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Call Routing > HTTP Profile**.

**Step 2**
Enter a **Name** for the service.

**Step 3**
Enter the **User Name** and **Password** for this HTTP connection. The user name does not have to be a configured end user in Cisco Unified Communications Manager, but the user name and password must match the values that are configured in the APIC-EM Controller.

**Step 4**
In the **Web Service Root URI** text box, enter the IP address or fully qualified domain name of the APIC-EM service.

**Step 5**
Configure any remaining fields in the HTTP Profile window. For help with the fields and their options, refer to the online help.

**Step 6**
Click **Save**.

**What to do next**

Enable External QoS Service for System, on page 544

Enable External QoS Service for System

Use this procedure to configure Cisco Unified Communications Manager to use an external service for QoS management. You must enable this service parameter in order to use an APIC-EM controller for QoS.

**Before you begin**

Configure HTTPS Connection to APIC-EM Controller, on page 544
Procedure

**Step 1**
From Cisco Unified CM Administration, choose System > Service Parameters.

**Step 2**
From the Server drop-down list, select the publisher node.

**Step 3**
From the Service drop-down list, select Cisco CallManager.

**Step 4**
Set the value of the External QoS Enabled service parameter to True.

**Step 5**
Click Save.

*Note* To use the APIC-EM to manage call flows for devices, you must also enable external QoS within the SIP Profile for the device.

**What to do next**
Configure External QoS Service at SIP Profile Level, on page 545

---

**Configure External QoS Service at SIP Profile Level**

If you have enabled the External QoS Enabled clusterwide service parameter, use this procedure to enable external QoS for SIP devices that use this SIP Profile.

*Note*
External QoS must be enabled at both the system level and in the SIP Profile to use the APIC-EM to manage QoS.

**Before you begin**
Enable External QoS Service for System, on page 544

**Procedure**

**Step 1**
In Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.

**Step 2**
Do one of the following:
- Click Find and select an existing SIP Profile.
- Click Add New to create a new SIP Profile.

**Step 3**
Check the Enable External QoS check box. This check box must be checked for phones that use this SIP Profile to use the APIC-EM Controller to manage QoS.

**Step 4**
Complete the remaining fields in the SIP Profile Configuration window. For help with the fields and their settings, see the online help.

**Step 5**
Click Save.
Assign SIP Profile to Phones

Use this procedure if you want to assign the external QoS-enabled SIP Profile that you created to a phone.

**Tip**

Use the Bulk Administration Tool to update the SIP Profile for a large selection of phones in a single operation. For details, see the *Bulk Administration Guide for CiscoUnifiedCommunicationsManager*.

**Before you begin**

Assign SIP Profile to Phones, on page 546

**Procedure**

**Step 1**
In Cisco Unified CM Administration, choose Device > Phone.

**Step 2**
Click Find and select an existing phone.

**Step 3**
From the SIP Profile drop-down list box, select the SIP Profile that you updated for phones that will use the APIC-EM Controller to manage traffic.

**Step 4**
Complete any remaining fields in the Phone Configuration window. For help with the fields and their settings, see the online help.

**Step 5**
Click Save.
Configure Call Control Discovery

- Call Control Discovery Overview, on page 547
- Call Control Discovery Prerequisites, on page 547
- Call Control Discovery Configuration Task Flow, on page 547
- Call Control Discovery Interactions and Restrictions, on page 554

Call Control Discovery Overview

Use Call Control Discovery (CCD) to advertise Unified Communications Manager information along with other key attributes, such as directory number patterns. Other call control entities that use the Service Advertisement Framework (SAF) network can use the advertised information to dynamically configure and adapt their routing operations. All entities that use SAF advertise their directory number patterns along with other key information. Other remote call control entities can learn the information from this broadcast and adapt the routing operations of the call.

Call Control Discovery Prerequisites

- SAF-enabled SIP or H.323 intercluster (non-gatekeeper controlled) trunks
- Remote call control entities that support and use the SAF network; for example, other Unified Communications Manager or Cisco Unified Communications Manager Express servers
- Cisco IOS routers that are configured as SAF forwarders

Call Control Discovery Configuration Task Flow

Before you begin

- Review Call Control Discovery Prerequisites, on page 547.
## Call Control Discovery Configuration Task Flow

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>See the documentation that supports your Cisco IOS router. Cisco Feature Navigator (<a href="http://www.cisco.com/go/cfn">http://www.cisco.com/go/cfn</a>) allows you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform.</td>
<td>Configure a Cisco IOS router as the SAF forwarder.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure SAF Security Profile, on page 549</td>
<td>Configure the SAF security profile for the SAF forwarder to provide a secure connection between the SAF forwarder and Unified Communications Manager.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure SAF Forwarders, on page 550</td>
<td>Configure the SAF forwarders, which are Cisco IOS routers configured for SAF. They notify the local cluster when remote call-control entities advertise their hosted DN patterns. In addition, the SAF forwarder receives publishing requests from the local cluster for each configured and registered trunk that is configured; the publishing request contains the Hosted DN patterns for the Cisco Unified Communications Manager, the PSTN failover configuration, the listening port for the trunk, and, for SIP trunks, the SIP route header field, which contains a URI for the trunk.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure SIP or H.323 Intercluster Trunks, on page 550</td>
<td>Configure SIP or H.323 intercluster (non-gatekeeper controlled) trunks for SAF support. The local cluster uses SAF-enabled trunks that are assigned to the CCD requesting service to route outbound calls to remote call-control entities that use the SAF network.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure Hosted DN Groups, on page 551</td>
<td>Configure hosted DN groups, which are collections of hosted DN patterns. After you assign a hosted DN group to the CCD advertising service, the CCD advertising service advertises all the hosted DN patterns that are a part of the hosted DN group. You can assign only one hosted DN group per CCD advertising service.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure Hosted DN Patterns, on page 551</td>
<td>Configure hosted DN patterns, which are directory number patterns that belong to Unified Communications Manager; the CCD advertising service advertises these patterns to other remote call-control entities that use the SAF network. You associate these patterns</td>
</tr>
</tbody>
</table>
### Configure SAF Security Profile

Configure the SAF security profile for the SAF forwarder to provide a secure connection between the SAF forwarder and Unified Communications Manager.

**Tip**

Use the same username and password that you entered on the router (SAF forwarder).

**Before you begin**

Configure a Cisco IOS router as the SAF forwarder. (See the Cisco Feature Navigator at [http://www.cisco.com/go/cfn](http://www.cisco.com/go/cfn).)

**Procedure**

**Step 1**

From Cisco Unified CM Administration, choose *Advanced Features > SAF > SAF Security Profile.*
Configure SAF Forwarders

Configure the SAF forwarders, which are Cisco IOS routers configured for SAF. They notify the local cluster when remote call-control entities advertise their hosted DNs patterns. In addition, the SAF forwarder receives publishing requests from the local cluster for each configured and registered trunk that is configured; the publishing request contains the Hosted DN patterns for the Cisco Unified Communications Manager, the PSTN failover configuration, the listening port for the trunk, and, for SIP trunks, the SIP route header field, which contains a URI for the trunk.

Tip
If more than one node appears in the Selected Cisco Unified Communications Managers pane, append @ to the client label value; otherwise, errors can occur if each node uses the same client label to register with the SAF forwarder.

Procedure

Step 1
From Cisco Unified CM Administration, choose Advanced Features > SAF > SAF Forwarder.
Step 2
Configure the fields on the SAF Forwarder Configuration window. See the online help for more information about the fields and their configuration options.
Step 3
Click Save.

Configure SIP or H.323 Intercluster Trunks

Configure SIP or H.323 intercluster (non-gatekeeper controlled) trunks for SAF support. The local cluster uses SAF-enabled trunks that are assigned to the CCD requesting service to route outbound calls to remote call-control entities that use the SAF network.

Procedure

Step 1
From Cisco Unified CM Administration, choose Device > Trunk.
Step 2
Click Add New.
Step 3
Perform one of the following tasks:

- For SIP trunks:
  a. From the Trunk Service Type Type drop-down list, choose Call Control Discovery. You cannot change the trunk service type after you select it from the drop-down list.
  b. Click Next.
Configure Advanced Call Handling

Configure Hosted DN Groups

Configure the fields on the **Trunk Configuration** window. See the online help for more information about the fields and their configuration options.

- For intercluster (non-gatekeeper controlled) trunks:
  a. Click **Next**.
  b. Check the **Enable SAF** check box.
  c. Configure the other fields on the **Trunk Configuration** window. See the online help for more information about the fields and their configuration options.

Step 4 Click **Save**.

---

**Configure Hosted DN Groups**

Configure hosted DN groups, which are collections of hosted DN patterns. After you assign a hosted DN group to the CCD advertising service, the CCD advertising service advertises all the hosted DN patterns that are a part of the hosted DN group. You can assign only one hosted DN group per CCD advertising service.

**Procedure**

Step 1 From Cisco Unified CM Administration, choose **Call Routing > Call Control Discovery > Hosted DN Group**.

Step 2 Configure the fields on the Hosted DN Groups Configuration window. See the online help for more information about the fields and their configuration options.

Step 3 Click **Save**.

---

**Configure Hosted DN Patterns**

Configure hosted DN patterns, which are directory number patterns that belong to Unified Communications Manager; the CCD advertising service advertises these patterns to other remote call-control entities that use the SAF network. You associate these patterns with hosted DN groups, which allow you to easily associate multiple patterns to a CCD advertising service.

**Procedure**

Step 1 From Cisco Unified CM Administration, choose **Call Routing > Call Control Discovery > Hosted DN Patterns**.

Step 2 Configure the fields on the **Hosted DN Patterns Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 3 Click **Save**.
Configure the Advertising Service

Configure the call control discovery advertising service, which allows Unified Communications Manager to advertise the hosted DNs for the cluster and the PSTN failover configuration to remote call-control entities that use the SAF network.

Procedure

Step 1
From Cisco Unified CM Administration, choose Call Routing > Call Control Discovery > Advertising Service.

Step 2
Configure the fields in the Advertising Service Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 3
Click Save.

Configure the Partition for Call Control Discovery

Configure a call control discovery partition to ensure that the learned patterns are inserted into digit analysis under this partition.

Note
The CCD partition does not appear under Call Routing > Class of Control > Partition in Cisco Unified Communications Manager Administration.

Procedure

Step 1
From Cisco Unified CM Administration, choose Call Routing > Call Control Discovery > Partition.

Step 2
Configure the fields in the Call Control Discovery Partition Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 3
Click Save.

Configure the Requesting Service

Caution
Updating the Learned Pattern Prefix or Route Partition fields can affect system performance. To avoid system performance issues, we recommend that you update these fields during off-peak hours.

To ensure that your local cluster can detect advertisements from the SAF network, configure one call control discovery requesting service to listen for advertisements from remote call control entities that use the SAF network. In addition, the CCD requesting service ensures that learned patterns are inserted into the digit analysis.
Procedure

Step 1
From Cisco Unified CM Administration, choose Call Routing > Call Control Discovery > Requesting Service.

Step 2
Configure the fields in the Requesting Service Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 3
Click Save.

Configure your remote call control entity to use the SAF network. (See the documentation for your remote call control entity.)

Block Learned Patterns

Block learned patterns that remote call control entities send to the local Unified Communications Manager. Perform this procedure on learned patterns that you no longer want to use.

Before you begin
Configure your remote call control entity to use the SAF network. See the documentation that supports your remote call control entity.

Procedure

Step 1
From Cisco Unified CM Administration, choose Call Routing > Call Control Discovery > Block Learned Patterns.

Step 2
Click Add New.

Step 3
Configure one of the following fields:

- In the Learned Pattern field, enter the exact learned pattern that you want to block. You must enter the exact pattern that you want Cisco Unified Communications Manager to block.
- In the Learned Pattern Prefix field, enter the prefix to block a learned pattern based on the prefix that is prepended to the pattern.

Example:
For Learned Pattern, enter 235XX to block 235XX patterns.

Example:
For Learned Pattern Prefix, enter +1 to block patterns that use +1.

Step 4
In the Remote Call Control Entity field, enter the name of the remote call control entity that advertises the pattern that you want to block.

Step 5
In the Remote IP field, enter the IP address for the remote call control entity where you want to block the learned pattern.

Step 6
Click Save.
Call Control Discovery Interactions and Restrictions

Call Control Discovery Interactions

Table 81: Call Control Discovery Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLF Subscriptions</td>
<td>For a user to subscribe BLF status of a SAF learned pattern, Unified Communications Manager sends a SIP subscribe message over a SIP trunk to the remote cluster. This functionality is supported with only SAF-enabled SIP trunks.</td>
</tr>
<tr>
<td>Bulk Administration Tool</td>
<td>In the Bulk Administration Tool, you can import and export the configuration for SAF security profiles, SAF forwarder, CCD advertising service, CCD requesting service, hosted DN groups, and hosted DN patterns.</td>
</tr>
<tr>
<td>Call Detail Records</td>
<td>Unified Communications Manager supports redirecting onBehalfOf as SAFCCDRRequestingService with a redirection reason as SS_RFR_SAF_CCD_PSTNFAILOVER, which indicates that the call is redirected to a PSTN failover number.</td>
</tr>
<tr>
<td>Incoming Called Party Settings</td>
<td>The H.323 protocol does not support the international escape character +. To ensure that the correct DN patterns are used with SAF and call control discovery for inbound calls over H.323 gateways or trunks, you must configure the incoming called party settings in the service parameter, device pool, H.323 gateway, or H.323 trunk windows; that is, configure the incoming called party settings to ensure that when a inbound call comes from a H.323 gateway or trunk, Unified Communications Manager transforms the called party number back to the value that was originally sent over the trunk or gateway. For example, a caller places a call to +19721230000 to Unified Communications Manager A. Unified Communications Manager A receives +19721230000 and transforms the number to 55519721230000 before sending the call to the H.323 trunk. In this case, your configuration indicates that the international escape character + should be stripped and 555 should be prepended for calls of International type. For this inbound call from the trunk, Unified Communications Manager B receives 55519721230000 and transforms the number back to +19721230000 so that digit analysis can use the value as it was sent by the caller. In this case, your configuration for the incoming called party settings indicates that you want 555 to be stripped and +1 to be prepended to called party numbers of International type.</td>
</tr>
<tr>
<td>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>-----------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Digest Authentication</td>
<td>Unified Communications Manager uses digest authentication (without TLS) to authenticate to the SAF forwarder. When Unified Communications Manager sends a message to the SAF forwarder, Unified Communications Manager computes the SHA1 checksum and includes it in the MESSAGE-INTEGRITY field in the message.</td>
</tr>
<tr>
<td>QSIG</td>
<td>The QSIG Variant and ASN.1 ROSE OID Encoding settings in the <strong>H.323 Configuration</strong> window are advertised by the CCD advertising service. These settings affect decoding of QSIG messages for inbound tunneled calls; for call control discovery, they do not affect outgoing calls. The remote call-control entity determines whether QSIG tunneling is required for outgoing calls over H.323 trunks. If the remote call-control entity advertises that QSIG tunneling is required, the QSIG message is tunneled in the message of the outgoing call, even if the <strong>H.323 Configuration</strong> window in Cisco Unified CM Administration indicates that QSIG support is not required.</td>
</tr>
</tbody>
</table>

**Call Control Discovery Restrictions**

All clusters are limited to advertised or learned routes within the same autonomous system (AS).
CHAPTER 76

Configure External Call Control

- External Call Control Overview, on page 557
- External Call Control Prerequisites, on page 557
- External Call Control Configuration Task Flow, on page 558
- External Call Control Interactions and Restrictions, on page 564

External Call Control Overview

External call control lets an adjunct route server make call routing decisions for Unified Communications Manager by using the Cisco Unified Routing Rules Interface. When you configure external call control, Unified Communications Manager issues a route request that contains the calling party and called party information to the adjunct route server. That server receives the request, applies appropriate business logic, and returns a route response that instructs your system on how to route the call and any additional call treatment to apply.

The adjunct router influences how your system allows, diverts, or denies calls; modifies calling and called party information; plays announcements to callers; resets call history so that adjunct voicemail and IVR servers can properly interpret calling and called party information; and logs reason codes that indicate why calls were diverted or denied.

External call control provides the following functions:

- Best Quality Voice Routing—The adjunct route server monitors network link availability, bandwidth usage, latency, jitter, and MOS scores to ensure that calls are routed through voice gateways that deliver the best voice quality to all call participants.
- Least Cost Routing—The adjunct route server is configured with carrier contract information such as local access and transport area (LATA) and inter-LATA rate plans, trunking costs, and burst utilization costs to ensure that calls are routed over the most cost effective links.
- Ethical Wall—The adjunct route server is configured with corporate policies that determine reachability, for example, whether user 1 is allowed to call user 2.

External Call Control Prerequisites

This feature requires the Cisco Unified Routing Rules XML Interface, which directs your system on how to handle calls.
For more information, see the *Cisco Unified Routing Rules Interface Developers Guide* (CURRI documentation) at https://developer.cisco.com.

# External Call Control Configuration Task Flow

**Before you begin**

- Review **External Call Control Prerequisites**, on page 557.

## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>Configure a Calling Search Space for External Call Control</strong>, on page 559</td>
<td>Configure a calling search space for your system to use when the route server sends a divert obligation. A calling search space comprises an ordered list of route partitions that you assign to devices. Calling search spaces determine the partitions that calling devices search when they attempt to complete a call.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>Configure an External Call Control Profile</strong>, on page 560</td>
<td>Configure an external call control profile to provide the URIs for the adjunct route server, a calling search space that is used for diverting calls, a timer that indicates how long your system waits for a response from the adjunct route server, and so on.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>Assign a Profile to a Translation Pattern</strong>, on page 560</td>
<td>For the translated patterns that you want to use with external call control, assign an external call control profile to the pattern. When a call occurs that matches the translation pattern, your system immediately sends a call routing query to an adjunct route server, and the adjunct route server directs your system on how to handle the call.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>(Optional) <strong>Import the Route Server Certificate into the Trusted Store</strong>, on page 561</td>
<td>If the route server uses HTTPS, import the certificate for the route server into the trusted store on your system node. You must perform this task on each node in the cluster that can send routing queries to the route server. If you use HTTPS for the primary or secondary web service URIs in the external call control profile, your system uses certificates to mutually authenticate through a TLS connection to the configured adjunct route servers.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>(Optional) <strong>Export the Self-Signed Certificate to the Route Server</strong>, on page 562</td>
<td>If the route server uses HTTPS, export the Cisco Unified Communications Manager self-signed...</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td>certificate to the route server. You must perform this task for each node in the cluster that can send routing queries to the route server. To ensure that the primary and redundant route servers can authenticate with Cisco Unified Communications Manager through HTTPS, you must generate a self-signed certificate that you can import to each adjunct route server that sends directives to your system. Perform this procedure for each node in the cluster that can contact the primary and redundant adjunct route server.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Step 6** (Optional) Configure the Chaperone Function, on page 562

Configure chaperone functionality if your routing rules from the route server state that a chaperone must monitor or record a call. A chaperone is a designated phone user who can announce company policies in the call, monitor the call, and record the call.

**Step 7** (Optional) Configure Customized Announcements, on page 563

Follow this procedure if your routing rules require that an announcement is played for some calls and you do not want to use the Cisco-provided announcements.

---

**Configure a Calling Search Space for External Call Control**

Configure a calling search space for your system to use when the route server sends a divert obligation. A calling search space comprises an ordered list of route partitions that you assign to devices. Calling search spaces determine the partitions that calling devices search when they attempt to complete a call.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Procedure</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified CM Administration, choose Call Routing &gt; Class of Control &gt; Calling Search Space.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click Add New.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>In the Name field, enter a name. Ensure that each calling search space name is unique to the system. The name can include up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>In the Description field, enter a description. The description can include up to 50 characters in any language, but it cannot include double-quotes (&quot;), percentage sign (%), ampersand (&amp;), back-slash (), or angle brackets (&lt;&gt;).</td>
</tr>
</tbody>
</table>
| **Step 5** | From the Available Partitions drop-down list, perform one of the following steps:  
  - For a single partition, select that partition.  
  - For multiple partitions, hold down the Control (CTRL) key, then select the appropriate partitions. |
Configure an External Call Control Profile

Configure an external call control profile to provide the URIs for the adjunct route server, a calling search space that is used for diverting calls, a timer that indicates how long your system waits for a response from the adjunct route server, and so on.

Before you begin
Configure a Calling Search Space for External Call Control, on page 559

Procedure

Step 1  From Cisco Unified CM Administration, choose Call Routing > External Call Control Profile.
Step 2  Perform one of the following tasks:
  • To modify the settings for an existing external call control profile, enter search criteria, click Find, and then choose an existing external call control profile from the resulting list.
  • To add a new external call control profile, click Add New.
Step 3  Configure the fields on the External Call Control Profile Configuration window. See the online help for more information about the fields and their configuration options.
Step 4  Click Save.

What to do next
Assign a Profile to a Translation Pattern, on page 560
Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Translation Pattern.

Step 2 Perform one of the following tasks:
- To modify the settings for an existing translated pattern, enter search criteria, click Find, and then choose an existing translated pattern from the resulting list.
- To add a new translated pattern, click Add New.

Step 3 From the External Call Control Profile drop-down list, choose the external call control profile that you want to assign to the pattern.

Step 4 Configure other fields as needed in the Translation Pattern Configuration window. See the online help for more information about the fields and their configuration options.

Step 5 Click Save.

What to do next
(Optional) Import the Route Server Certificate into the Trusted Store, on page 561

Import the Route Server Certificate into the Trusted Store

If the route server uses HTTPS, import the certificate for the route server into the trusted store on your system node. You must perform this task on each node in the cluster that can send routing queries to the route server. If you use HTTPS for the primary or secondary web service URIs in the external call control profile, your system uses certificates to mutually authenticate through a TLS connection to the configured adjunct route servers.

Before you begin
Assign a Profile to a Translation Pattern, on page 560

Procedure

Step 1 From Cisco Unified Operating System Administration, choose Security > Certificate Management.

Step 2 Click Upload Certificate.

Step 3 In the Upload Certificate popup window, click CallManager-trust from the Certificate Name drop-down list, and browse to the certificate for the adjunct route server.

Step 4 After the certificate appears in the Upload File field, click Upload.

Step 5 (Optional) Perform this procedure again if your system can contact a redundant adjunct route server.

What to do next
Export the Self-Signed Certificate to the Route Server, on page 562
Export the Self-Signed Certificate to the Route Server

If the route server uses HTTPS, export the Cisco Unified Communications Manager self-signed certificate to the route server. You must perform this task for each node in the cluster that can send routing queries to the route server. To ensure that the primary and redundant route servers can authenticate with Cisco Unified Communications Manager through HTTPS, you must generate a self-signed certificate that you can import to each adjunct route server that sends directives to your system.

Perform this procedure for each node in the cluster that can contact the primary and redundant adjunct route server.

Before you begin
Import the Route Server Certificate into the Trusted Store, on page 561

Procedure

Step 1 From Cisco Unified Operating Administration, choose Security > Certificate Management.
Step 2 In the Certificate List window, click Generate New.
Step 3 From the Certificate Name drop-down list, choose CallManager.
Step 4 Click Generate New.
Step 5 From the Find and List Certificates window, choose the CallManager.pem certificate that you just created.
Step 6 After the certificate file data appears, click Download to download the certificate to a location that you can use for exporting the certificate to the adjunct route server.
Step 7 Export the certificate to each adjunct route server that sends directives.

What to do next
(Optional) Configure the Chaperone Function, on page 562

Configure the Chaperone Function

Configure chaperone functionality if your routing rules from the route server state that a chaperone must monitor or record a call. A chaperone is a designated phone user who can announce company policies in the call, monitor the call, and record the call.

Cisco Unified Communications Manager provides the following capabilities to support chaperone functionality, as directed by the adjunct route server:

- Redirect an incoming call to a chaperone, hunt group, or a list of chaperones.
- Provide a chaperone with the ability to record a call.

When the chaperone is connected to the caller or when the chaperoned conference is established, the Record softkey or programmable line key (PLK) (depending on the phone model) is active on the phone so that the chaperone can invoke call recording. Call recording occurs for only the current call, and call recording stops when the current call ends. Messages that indicate the status of recording may display on the phone when the chaperone presses the recording softkey or PLK.
Before you begin

(Optional) Export the Self-Signed Certificate to the Route Server, on page 562

Procedure

Step 1  For phones on which you want to enable recording, set the Built-in-Bridge to On in the Phone Configuration window.

Step 2  Create a recording profile:
   a) Choose Device > Device Settings > Recording Profile.
   b) Create a Call Recording Profile for the phones that can record chaperoned conferences.

Step 3  Apply the recording profile to the line appearance.

Step 4  Add a SIP trunk to point to the recorder.

Step 5  Create a route pattern that points to the SIP trunk.

Step 6  Configure the following service parameters:
   a) Play Recording Notification Tone to Observed Target
   b) Play Recording Notification Tone to Observed Connected Target

Step 7  Assign the Standard Chaperone Phone softkey template to the phone that the chaperone uses.

Step 8  Perform the following steps from Call Routing > Directory Number for a new phone or from Device > Phone if the phone is already configured:
   a) Configure only one directory number (DN) for the chaperone phone.
   b) For the DN on the chaperone phone, choose Device Invoked Call Recording Enabled from the Recording Option drop-down list.
   c) For the DN on the chaperone phone, enter 2 for the Maximum Number of Calls setting, and enter 1 for the Busy Trigger setting.

Step 9  For Cisco Unified IP Phones that support the Record softkey, configure the Standard Chaperone Phone softkey template so that only the Conference, Record, and End Call softkeys display on the phone in a connected state.

Step 10  For Cisco Unified IP Phones that support the record programmable line key (PLK), configure the PLK in the Phone Button Template Configuration window.

Step 11  (Optional) If you have more than one chaperone in your cluster, add the chaperone DN to the chaperone line group that you plan to assign to the chaperone hunt list.

   This step ensures that an available chaperone monitors the call.

What to do next

(Optional) Configure Customized Announcements, on page 563

Configure Customized Announcements

Follow this procedure if your routing rules require that an announcement is played for some calls and you do not want to use the Cisco-provided announcements.
Tip
Do not use embedded spaces for the announcement identifier.

If other language locales are installed, you can upload other .wav files for this announcement to use with those locales.

Procedure

Step 1
From Cisco Unified CM Administration, choose Media Resources > Announcement.

Step 2
Perform one of the following tasks:
  • To add a new announcement:
    a) Click Add New.
    b) In the Announcement Identifier field, enter an announcement identifier.
    c) In the Description, enter a description of the announcement.
    d) From the Default Announcement drop-down list, choose a default Cisco-provided announcement if desired.
    e) Click Save.
  • To upload a custom .wav file for the announcement:
    a) Click Upload File.
    b) From the Locale drop-down list, choose the locale language for the announcement.
    c) Click Choose File, and then choose a .wav file to upload.
    d) Click Upload File.
    e) When the upload finishes, click Close to refresh the window and show the uploaded announcement.

External Call Control Interactions and Restrictions

External Call Control Interactions

Table 82: External Call Control Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Best Call Quality Routing</td>
<td>You can set up routing rules on the adjunct route server that determine which gateway to use for a call, taking voice quality into consideration. For example, gateway A provides the best voice quality, so it is used for the call. In this case, the adjunct route server monitors network link availability, bandwidth usage, latency, jitter, and mean opinion scores (MOS) to ensure that calls are routed through voice gateways that deliver the best voice quality to all call participants.</td>
</tr>
</tbody>
</table>
### External Call Control Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Detail Records</td>
<td>External Call Control functions can be displayed in call detail records; for example, the call detail record can indicate whether the adjunct route server permitted or rejected a call. In addition, the call detail record can indicate whether Cisco Unified Communications Manager blocked or allowed calls during which it did not receive a decision from the adjunct route server.</td>
</tr>
</tbody>
</table>
| Call Forward          | External Call Control intercepts calls at the translation pattern level, while Call Forward intercepts calls at the directory number level. External Call Control has a higher priority than Call Forward; for calls that invoke Call Forward, Cisco Unified Communications Manager sends a routing query to the adjunct route server if the translation pattern is assigned to an External Call Control profile. Call Forward is triggered only when the adjunct route server sends a Permit decision with a Continue obligation to the Cisco Unified Communications Manager.  
  **Note** The **Call Diversion Hop Count** service parameter that supports External Call Control and the **Call Forward Call Hop Count** service parameter that supports Call Forward are independent of each other. |
| Call Pickup           | When a phone user tries to pick up a call by using the Call Pickup feature, External Call Control is not invoked; Cisco Unified Communications Manager does not send a routing query to the adjunct route server for that portion of the call. |
| Chaperones            | A chaperone is a designated phone user who can announce company policies to the call, monitor the call, and record the call, if required. Chaperone restrictions exist so that the parties that are involved in the call cannot converse without the presence of the chaperone. |
| Cisco Unified Mobility| Cisco Unified Communications Manager allows the route decision from the adjunct route server for the following Cisco Unified Mobility features:  
  - Mobile Voice Access  
  - Enterprise Feature Access  
  - Dial-via-Office Reverse Callback  
  Cisco Unified Communications Manager does not send a routing query for the following Cisco Unified Mobility features:  
  - Cell pickup  
  - Desk pickup  
  - Session handoff |
| Conferences           | When a phone user creates a conference, External Call Control may be invoked for the primary call and consultative call. |
### Directory Numbers

When you configure directory numbers as four- or five-digit extensions (enterprise extensions), you need to configure two translation patterns if on-net dialing supports four or five digits. One translation pattern supports globalizing the calling and called numbers, and a second translation pattern supports localizing the calling and called numbers.

### Do Not Disturb

By default, the DND setting for the user takes effect when the user rule on the adjunct route server indicates that the adjunct route server sent a continue obligation. For example, if the adjunct route server sends a continue obligation, and the user has DND-R enabled, Cisco Unified Communications Manager rejects the call.

### Emergency Call Handling

**Caution**  
We strongly recommend that you configure a very explicit set of patterns for emergency calls (for example, 911 or 9.11) so that the calls route to their proper destination (for example, to Cisco Emergency Responder or a gateway) without having to contact the route server for instructions on how to handle the call.

### Transfer

When a phone user transfers a call, External Call Control may be invoked for both the primary call and consultative call. However, Cisco Unified Communications Manager cannot enforce any routing rules from the adjunct route server between the party that transfers and the target of the transfer.

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### External Call Control Restrictions

**Table 83: External Call Control Restrictions**

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| Adding Parties     | The chaperone cannot use the phone to add parties to a conference after the conference begins, because the call must be put on hold for the chaperone to add parties.  
The other parties on the conference may add additional parties to the conference. The configuration for the **Advanced Ad Hoc Conference Enabled** service parameter, which supports the Cisco CallManager service, determines whether other parties can add participants to the conference. If the service parameter is set to **True**, other parties can add participants to the conference. |
| Call Transfer      | The chaperone cannot use the phone to transfer the conference call to another party. |
| Conference Log Out | When the chaperone leaves the conference, the entire conference ends.         |
| Conference Softkey | After the chaperone creates a conference, the **Conference** softkey, if available, is disabled on the phone. |
| Hold               | The chaperone cannot use the phone to put the conference call on hold.        |
| Recording          | If the chaperone starts recording before the feature makes a consultative call to the party that will join the conference, Cisco Unified Communications Manager suspends recording while the chaperone makes the consultative call; recording resumes after the conference is established. |
Configure Call Queuing

• Call Queuing Overview, on page 567
• Call Queuing Prerequisites, on page 568
• Call Queuing Task Flow, on page 568
• Call Queuing Interactions and Restrictions, on page 576

Call Queuing Overview

Cisco Unified Communications Manager provides Call Queuing to place callers in a queue until hunt members are available to answer them. An administrator can set the default so callers receive an initial greeting announcement before the call is extended to an agent or the default can be changed so the initial announcement plays only after the caller is put in the queue followed by Music On Hold or Tone On Hold. If the caller remains in queue for a specified period of time, a secondary announcement is played at a configured interval until the call can be answered or until the maximum wait timer expires.

When an incoming call reaches the hunt pilot, the following functions are provided:

• A caller may be connected to an initial customizable greeting announcement before proceeding.
• If one or more line members are logged in to the hunt pilot and are in an idle state, and if no calls are queued, the call is extended to the line member that has been idle for the longest period of time.
• If no line members answer a call, that caller is not placed in queue. The call is routed to a new destination or disconnected, based on the setting When no hunt members answer, are logged in, or registered.
• If a line member does not answer a queue-enabled call, that line member is logged off the hunt group only if the setting Automatically Logout Hunt Member on No Answer is selected in the Line Group setting window.
• Calls are placed in queue only if all members are busy.
• A caller who is waiting in queue may hear Music On Hold and a repeating (customizable) periodic announcement.
• After a line member becomes idle, the caller with the longest wait time across multiple hunt groups is extended to the idle line member. If the idle line member does not answer the call, the caller is returned to the previous position in the queue.
• If a queued call exceeds its maximum wait time or the maximum number of callers allowed in queue is exceeded, the call can be routed to an alternate number or it can be disconnected, depending on how the hunt pilot is configured. The alternate number can be one of the following:
  • A hunt pilot DN with queuing either enabled or disabled
  • A voicemail DN
• A line DN
• A shared DN

• Line members can display the queue status of their queue-enabled hunt pilots. The queue status display provides the following types of information:
  • Hunt pilot pattern
  • Number of queued callers on each hunt pilot
  • Longest waiting time

Call queuing works in conjunction with existing hunt pilots, but there are no changes in the behavior of the hunting operation for either queueing or nonqueueing hunt pilots. Hunt pilots that have call queuing enabled provide the following features:

  • Queuing-enabled hunt pilot calls can only be received by line members one call at a time. Two queuing-enabled hunt pilot calls cannot be offered to a line member. A line member can receive calls directly to the DN or from non-queuing hunt pilots.
  • Line members who do not answer calls that are routed by hunt pilots are automatically logged out. A line member is automatically logged out of a device if the line member receives a queueing-enabled hunt pilot call and does not answer the call before timeout occurs. In the case of a shared-line deployment, all devices configured with the same shared line are logged out. You can configure this behavior from the Line Group setting window by selecting Automatically Logout Hunt Member on No Answer. Line members are logged out only if this check box is checked.

For information about Call Queuing monitoring or announcements monitoring, see *Cisco Unified Real Time Monitoring Tool Administration Guide*.

You can configure the inbound calls to change to the connected call state before playing the queuing announcement while the call is extended to a hunt member in the queueing-enabled hunt pilot.

**Call Queuing Prerequisites**

- Cisco IP Voice Media Streaming (IPVMS) Application, which should be activated on at least one node in the cluster
- Cisco CallManager service that is running on at least one server in the cluster
- Cisco RIS Data Collector service that is running on the same server as the Cisco CallManager service
- Cisco Unified Communications Manager Locale Installer, if you want to use non-English phone locales or country-specific tones

**Call Queuing Task Flow**

*Before you begin*

**Configure Announcements**

Cisco Unified Communications Manager allows you to:
• use the existing Cisco-provided announcements,
• change the message or tone that you want an announcement to play,
• insert custom announcement .wav files,
• assign the locale for the announcement,
• change the description for the announcement,
• change the message or tone that you want an announcement to play.

Feature announcements are used by specific features such as Music On Hold (MoH) in association with Hunt Pilot call queuing or External Call Control.

There are up to 50 feature announcements available. These announcements can be Cisco-provided audio files or uploaded custom .wav files.

All custom announcement .wav files must be uploaded to all servers in the cluster.

Procedure

Step 1  In Cisco Unified Communications Manager, select Media Resources > Announcements. The Find and List Announcements window displays.

Step 2  Select a hyperlink to the announcement you want to use.
Example: Hyperlink—Wait_In_Queue_Sample You can edit the announcement description or choose a customized announcement if uploaded.

Step 3  To upload a .wav file to use as a custom announcement, click Upload File. The Upload File window opens.

Step 4  In the Upload File window, choose the locale, enter the filename or browse to select the .wav file, and click Upload File.
The upload process begins, and may take a few minutes depending on the file. The Status is updated after processing is complete.

Step 5  Click Close to close the upload window. The Announcement Configuration window refreshes to update the uploaded file status.

Step 6  To play the customized announcement, ensure that the Enable check box is checked in the Announcement by Locale pane in the Announcements Configuration window.

Step 7  After you make the changes in the Announcements Configuration window, click Save.

What to do next

You must upload the announcement on each node in the cluster, because the announcement files are not propagated between servers in a cluster. Browse to Cisco Unified Communications Manager Administration on each server in the cluster and repeat the upload process.
Configure Music On Hold

You can configure Music On Hold (MoH) to play an optional initial greeting announcement when a caller is first put on hold and also to play a periodic repeating announcement. These announcements can use one of the Cisco-provided audio files or a file that is uploaded into the system.

Perform the following procedure to add or update a Music On Hold audio source, to associate an existing audio source with an audio stream number, or to upload a new custom audio source.

Procedure

**Step 1**
From the Cisco Unified Communications Manager, choose **Media Resources > Music On Hold Audio Source**.

The **Find and List Music On Hold Audio Sources** window appears.

**Step 2**
To add a new Music On Hold audio source, click **Add New**. To update a Music On Hold audio source, locate a specific Music On Hold audio source. Based on the search criteria you specify, the system displays search results for the record that matches all the criteria.

**Step 3**
Enter the appropriate settings, as described in **Audio Source Fields for Music On Hold**, on page 570.

**Step 4**
Click **Save**.

The list box at the bottom of the window shows the new Music On Hold audio source. The MOH Audio Source File Status pane shows the MOH audio translation status for the added source.

Audio Source Fields for Music On Hold

**Table 84: Music On Hold Audio Source Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MOH Audio Stream Number</td>
<td>Use this field to choose the stream number for this MOH audio source. Click the drop-down arrow and choose a value from the list. For existing MOH audio sources, the value appears in the MOH Audio Source title.</td>
</tr>
<tr>
<td>MOH Audio Source File</td>
<td>Use this field to choose the file for this MOH audio source. Click the drop-down arrow and choose a value from the list.</td>
</tr>
<tr>
<td>MOH Audio Source Name</td>
<td>Enter a unique name in this field for the MOH audio source. This name includes up to 50 valid characters, such as letters, numbers, spaces, dashes, dots (periods), and underscores.</td>
</tr>
<tr>
<td>Allow Multicasting</td>
<td>Check this check box to specify that the selected MOH audio source allows multicasting.</td>
</tr>
</tbody>
</table>
### Field | Description
---|---
MOH Audio Source File Status | This pane displays the following information about the source file for the selected MOH audio source:  
- InputFileName  
- ErrorCode  
- ErrorText  
- DurationSeconds  
- DiskSpaceKB  
- LowDateTime  
- HighDateTime  
- OutputFileList  
- MOH Audio Translation completion date  
**Note** | OutputFileList includes information on ULAW, ALAW, G.729, and Wideband wav files and status options.

### Table 85: Announcement Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Initial Announcement | Choose an initial announcement from the drop-down list.  
**Note** | To select MoH with no initial announcement, choose the Not Selected option.  
Click the View Details link to view the following Initial Announcement information:  
- Announcement Identifier  
- Description  
- Default Announcement  
**Note** | • Played by MOH server only when the Audio Source has “Allow Multi-casting” unchecked and “Initial Announcement Played” set to 'Only for queued calls'.  
• Played by ANN if “Allow Multi-casting” is checked or if “Initial Announcement Played” is set to 'Always'.}
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Initial Announcement Played  | Choose one of the following to determine when to play the initial announcement:  
  • Play announcement before routing to Hunt Member  
  • Play announcement if call is queued |
| Periodic Announcement        | Choose a periodic announcement from the drop-down list.  
  **Note** To select MoH with no periodic announcement, choose the **Not Selected** option.  
  Click the **View Details** link to view the following Periodic Announcement information:  
  • Announcement Identifier  
  • Description  
  • Default Announcement  
  **Note** The MOH server always plays the periodic announcement regardless of other settings. |
| Periodic Announcement Interval| Enter a value (in seconds) that specifies the periodic announcement interval. Valid values are 10 to 300. The default value is 30. |
| Locale Announcement          | Locale Announcement depends upon the locale installation package that has been installed.  
  **Note**  
  • Prompts played by MOH will use the setting for Locale Announcement.  
  • Prompts played by ANN will use the User Locale of the calling party. |
**Table 86: Music On Hold Audio Sources**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>(list of MoH audio sources)</td>
<td>This list box shows the MOH audio source that you add. Select the audio stream number of an MOH audio source to configure that MoH audio source. Audio source ID is an ID that represents an audio source in the Music On Hold server. The audio source can include either a file on a disk or a fixed device from which a source stream Music On Hold server obtains the streaming data. An MOH server can support up to 51 audio source IDs. Each audio source, represented by an audio source ID, can stream as unicast and multicast mode, if needed. <strong>Note</strong> If you select &lt;None&gt;, the system default MoH audio source service parameter (Default Network Hold MoH Audio Source ID) is used for the MoH audio source.</td>
</tr>
<tr>
<td>Upload File</td>
<td>To upload an MOH audio source file that does not appear in the drop-down list, click Upload File. In the <strong>Upload File</strong> window, either enter the path of an audio source file or navigate to the file by clicking <strong>Browse</strong>. After you locate the audio source file, click the <strong>Upload File</strong> button to complete the upload. After the audio file gets uploaded, the Upload Result window displays the result of the upload. Click <strong>Close</strong> to close this window. <strong>Note</strong> When you upload a file, the file is uploaded to the Cisco Unified Communications Manager server and performs audio conversions to create codec-specific audio files for MOH. Depending on the size of the original file, processing may take several minutes to complete. <strong>Note</strong> Uploading an audio source file to an MOH server uploads the file only to one MOH server. You must upload an audio source file to each MOH server in a cluster by using Cisco Unified Communications Manager Administration on each server. MOH audio source files do not automatically propagate to other MOH servers in a cluster.</td>
</tr>
</tbody>
</table>
Configure Hunt Pilot Queuing

When a hunt pilot has more calls distributed through the call distribution feature than its hunt members can handle at any given time, call queuing holds these calls in a queue until they can be answered.

When queuing is enabled, both Forward Hunt No Answer and Forward Hunt Busy are automatically disabled. Conversely, if Forward Hunt No Answer or Forward Hunt Busy is enabled, queuing is automatically disabled.

Procedure

Step 1
In Cisco Unified Communications Manager Administration, select Call Routing > Route/Hunt > Hunt Pilot to configure hunt pilots.

Step 2
Select the hunt pilot that you need to configure for Queuing.

Step 3
Navigate to the Queuing section of the Hunt Pilot Configuration window.

Step 4
Check the Queue Calls check box to enable queuing.

Step 5
Choose a Music On Hold (MoH) source from the drop-down list box to be used to play announcements and provide queue hold treatments.

The MoH source can be configured as unicast or multicast. The caller-side Media Resource Group List (MRGL) takes precedence for multicast or unicast.

If you do not select a source, the default Network Hold MoH/MoH Source and Announcements is used.

The MoH source announcement locale is used to determine the language used for the announcement. Only one type of language announcement can be played per hunt pilot.

Step 6
In the Maximum Number of Callers Allowed in Queue field, enter an integer value for the number of callers allowed in the queue for this hunt pilot.

The default value is 32. The field range is from 1 to 100.

Step 7
Choose one of the following options when the maximum number of callers in the queue is reached:

- If you want subsequent calls to be disconnected, select Disconnect the call.
- If you want subsequent calls to be routed to a secondary destination, select Route the call to this destination. Provide a specific device DN, shared line DN, or another hunt pilot DN.
- (Optional) You may also select Full Queue Calling Search Space from the drop-down list. Used to determine which partition to search when attempting to complete a call.

Step 8
In the Maximum Wait Time in Queue field, enter an integer value to set the maximum wait time, in seconds, in a queue.

The default value is 900 seconds. The field range is from 10 to 3600 seconds.

Step 9
Choose one of the following options when the maximum wait time is reached:

- If you want that call to be disconnected, select Disconnect the call.
- If you want that call to be routed to a secondary destination, select Route the call to this destination. Provide a specific device DN, shared line DN, or another hunt pilot DN.
- (Optional) You may also select Maximum Wait Time Calling Search Space from the drop-down list. Used to determine which partition to search when attempting to complete a call.

Step 10
When no line members are logged in or registered at the time of an incoming call, choose one of the following options:
• If you need that call to be disconnected, select **Disconnect the call**.
• If you need that call to be routed to a secondary destination, select **Route the call to this destination**. Provide a specific device DN, shared line DN, or another hunt pilot DN.
• (Optional) You may also select **No hunt members logged in or registered Calling Search Space** from the drop-down list. Used to determine which partition to search when attempting to complete a call.

**Step 11**  
Click **Save**.

---

### Automatically Logout Hunt Member on No Answer

**Procedure**

**Step 1**  
In Cisco Unified Communications Manager Administration, choose **Call Routing > Route/Hunt > Line Group** to configure line groups.

**Step 2**  
Choose the line group that you need to configure from the **Find and List Line Groups** window.

**Step 3**  
Navigate to the Hunt Options section of the **Line Group Configuration** window.

**Step 4**  
Ensure that the **Automatically Logout Hunt Member on No Answer** check box is checked.

**Step 5**  
Click **Save**.
## Call Queuing Interactions and Restrictions

### Call Queuing Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Rel1XX Options</td>
<td>If a call is routed to a queuing-enabled hunt pilot through SIP ICT, the SIP ICT uses the SIP profile that has SIP Rel1XX Options set to <strong>Send PRACK if 1XX contains SDP</strong>. As a result, the initial announcement is played to every call before the call is extended to the line member. The above existing interaction for SIP ICT does not apply if <strong>Connect Inbound Call before Playing Queuing Announcement</strong> checkbox is checked under <strong>Device Device Settings SIP Profile &gt; Trunk Specific Configuration</strong> in Cisco Unified CM Administration. If <strong>Connect Inbound Call before Playing Queuing Announcement</strong> checkbox is not checked the interaction for SIP ICT remains the same. However, it does not guarantee the initial announcement can always be heard by a caller from the PSTN side. The initial announcement will not be heard by a caller from the PSTN side if the PSTN provider doesn't open the voice path until a Connect message is received on the call.</td>
</tr>
</tbody>
</table>
| Hunt Pilots and Hunt Groups  | • The logoff notification functionality for hunt groups changes when Call Queuing is enabled for a hunt pilot. If Call Queuing is enabled for a hunt pilot, the Hunt Group Logoff Notification does not play when users log out of a hunt group or are logged off because they missed their turn in the queue.  
• If the hunt list has multiple line groups, these line groups must have the same setting for **Automatically Logout Hunt Member on No Answer**.  
• Hunt Pilot still queues calls, even when all hunt members are logged out. The line group members should not be added in more than one line group and even if they are added in second line groups, those second line groups should not be in the same Hunt list.  
• All hunt options must be set to Try Next Member, then Try Next Group in the hunt list. |
Call Queuing Restrictions

The following general restrictions apply to call queuing:

- H.323 Fast Start does not support Call Queuing.
- Queue status PLK is supported only with the following LCD display phones for both SCCP and SIP: 6921, 6941, 6945, 6961, 7911G, 7931G, 7942G, 7945G, 7962G, 7965G, 7975G, 8961, 8945, 8941, 9951, 9971, 7800 and 8800 series.
- Log Out of Hunt Groups (HLog) is not compatible with Cisco Extension Mobility Cross Cluster (EMCC); Call Queuing should not be deployed with EMCC.
- Unified Communications Manager does not support Unified Mobility with Call Queuing.
- In a H323 to SIP interworking scenario, the user may not hear initial announcement, MoH, periodic announcement or observe call failure in a native call queuing flow due to interworking delays. In such a scenario it is advised to use only SIP protocol.

Performance and Scalability for Hunt Pilots with Call Queuing

The following performance and scalability restrictions apply:

- A single Unified CM Cluster supports a maximum of 15,000 hunt list devices.
- A single Unified CM Subscriber supports a maximum of 100 hunt pilots with call queuing enabled per node
- Hunt list devices may be a combination of 1500 hunt lists with ten IP phones in each hunt list, 750 hunt lists with twenty IP phones in each hunt list, or similar combinations

Note

When using the broadcast algorithm for call coverage, the number of hunt list devices is limited by the number of busy hour call attempts (BHCA). Note that a BHCA of 10 on a hunt pilot pointing to a hunt list or hunt group containing 10 phones and using the broadcast algorithm is equivalent to 10 phones with a BHCA of 10.

- The maximum number of simultaneous callers in queue for each hunt pilot that you can configure ranges from 1 to 100 (default 32).
- The maximum wait time in queue for each hunt pilot that you can configure ranges from 0 to 3600 seconds (default 900). An increase in the number of hunt lists can require you to increase the dial plan initialization timer that is specified in the Unified Communications Manager service parameters. We recommend that you set the dial plan initialization timer to 600 seconds if you have 1500 hunt lists configured.
- We recommend having no more than 35 directory numbers for a single line group when using broadcast algorithms with call queuing. Additionally, the number of broadcast line groups depends on the busy hour call completion rate (BHCC). If there are multiple broadcast line groups in a Unified CM system, the number of maximum directory numbers in a line group must be less than 35. The number of busy hour call attempts (BHCA) for all the broadcast line groups should not exceed 35 calls set up per second.
CHAPTER 78

Configure Call Throttling

- Call Throttling Overview, on page 579
- Call Throttling Configuration, on page 580

Call Throttling Overview

Call Throttling allows your system to automatically throttle or deny new call attempts. The system takes this action when conditions cause users to experience a delay in the interval between going off hook and receiving a dial tone.

Some factors that can result in this delay are as follows:

- Heavy call activity
- Low CPU availability
- Routing loops
- Disk I/O limitations
- Disk fragmentation

The system uses the values that are specified in the call throttling parameters to determine a possible delay to dialtone and also to determine when conditions no longer require call throttling.

When throttling is necessary to prevent excessive delay to dialtone, the system enters a Code Yellow state and new call attempts are throttled (denied).

When the system calculates the delay to dialtone as being over the threshold that is configured in the call throttling service parameters, Unified Communications Manager rejects new calls. When call throttling activates, a user who attempts a new call receives a reorder tone and, depending on the phone model, may also receive a prompt on the phone display.

Call throttling effectively prevents the type of excessive delays that can cause a user to complain to the system administrator or question whether the system is down or the phone is broken. Your system constantly monitor the system to anticipate when such latency could occur.

When the delay to dialtone is within the guidelines of the call throttling service parameters, Unified Communications Manager stops throttling calls by exiting the Code Yellow state and new calls are again allowed.
Call Throttling Configuration

We recommend that you not modify call throttling parameters unless advised to do so by customer support.

Call throttling occurs automatically when your system detects conditions such as heavy call activity, low CPU availability, and disk fragmentation. The system automatically exits throttling when these conditions are fixed.

Call Throttling Service Parameters

<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Code Yellow Entry Latency</td>
<td>Defines the maximum allowable delay, in milliseconds, to handle System Diagnostic Layer (SDL) messages that are sent to Cisco Unified Communications Manager by the various devices in the system. This maximum also applies to internal messages that are received and sent by Cisco Unified Communications Manager for various activities, such as keepalive intervals and change notification. If the calculated average expected delay is more than the value that is specified in this service parameter, the system enters a Code Yellow state to initiate call throttling and stops accepting new calls.</td>
</tr>
</tbody>
</table>
| Code Yellow Exit Latency Calculation | Determines the acceptable percentage of Code Yellow Entry Latency to specify exit criteria for leaving the Code Yellow state (call throttling) after Cisco Unified Communications Manager initiates call throttling. The value that you specify in this parameter derives from a formula that uses the value in the Code Yellow Entry Latency parameter (the delay as measured in milliseconds). To arrive at a percentage, use the following formula: Code Yellow Entry Latency value multiplied by the Code Yellow Exit Latency value. For example:  
  - Code Yellow Entry Latency service parameter value: 20 milliseconds  
  - Code Yellow Exit Latency service parameter value: 40%  
  - Code Yellow Exit Latency value = 20 x 0.4 = 8 milliseconds, which means Cisco Unified Communications Manager exits Code Yellow state if the calculated message latency drops to 8 milliseconds or lower. To exit the Code Yellow state, Cisco Unified Communications Manager ensures that the average expected delay is less than the value of the Code Yellow Exit Latency. |
<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Code Yellow Duration</td>
<td>Specifies the number of minutes that a Cisco Unified Communications Manager system can remain in a Code Yellow state (call throttling). If this duration is met and the system is still in Code Yellow state, Cisco Unified Communications Manager enters a Code Red state, which indicates that Cisco Unified Communications Manager has remained in a Code Yellow state for an extended period and cannot recover. When Cisco Unified Communications Manager enters a Code Red state, the Cisco CallManager service restarts, which also produces a memory dump that may be helpful for analyzing the failure.</td>
</tr>
<tr>
<td>System Throttle Sample Size</td>
<td>Indicates the size of the sample, in seconds, that is used to calculate the average expected delay for Cisco Unified Communications Manager to handle an SDL message. For example, a sample size of 10 means that Cisco Unified Communications Manager must calculate a nonzero latency value for 10 consecutive seconds before it will calculate the average expected delay and compare it to the value in the CodeYellow Entry Latency parameter. You can disable call throttling by using this parameter.</td>
</tr>
</tbody>
</table>
CHAPTER 79

Calling Party Normalization

This chapter describes the Calling Party Normalization feature. Calling Party Normalization allows you to reformat the display of calling party numbers to a globalized version that includes prefixes such as international country codes or localize the calling party number to a localized version that displays on the called party's phone.

- Calling Party Normalization Overview, on page 583
- Calling Party Normalization Prerequisites, on page 584
- Calling Party Normalization Configuration Task Flow, on page 585
- Calling Party Normalization Interactions and Restrictions, on page 588

Calling Party Normalization Overview

Calling Party Normalization allows you to globalize and localize phone numbers so that the appropriate calling presentation displays on the phone. Calling Party Normalization enhances the dialing capabilities of some phones and improves callback functionality when a call is routed to multiple geographic locations. The feature allows you to map a global calling party number to its localized variant such that a phone can return a call without modifying the directory number in the call log directories on the phone.

Globalization of the Calling Party Number

By configuring a Calling Party Number Type and prefixes in Cisco Unified CM Administration, you can set Cisco Unified Communications Manager to reformat the calling party number that displays on the called phone to a globalized version that includes prefixes such as international country codes, thereby allowing the number to be dialed from anywhere in the world.

Cisco Unified Communications Manager uses various number patterns, such as route patterns or translation patterns, along with the value for the Calling Party Number Type to globalize a phone number. For example, you can configure Cisco Unified Communications Manager to take a localized German phone number of 069XXXXXXX with a Subscriber calling party number type and globalize it to +49 40 69XXXXXXX, which includes the German country code and city code.

For calls that are routed to multiple geographic locations, the different translation settings that are applied for each routing path can globalize the calling party number uniquely for each call path. Cisco Unified Communications Manager can also be configured such that the phone displays a localized calling party number on the phone screen and the globalized number in the call log directories on the phone. To ensure that the phone user does not need to edit the call log directory entry on the phone before placing a call, map the global calling party number to its local variant.
Localization of the Calling Party Number

For the final presentation of the calling party number, you can configure calling party transformation patterns for each calling party number type (National, International, Subscriber, and Unknown), and apply strip digits and prefix instructions specific to the calling party number type for that call. This allows Cisco Unified Communications Manager to reformat the calling party number such that the calling party number that displays on the called phone is a localized number that does not include unnecessary country codes and international access codes.

For example, assume an incoming number arrives from the PSTN with a globalized number of +49 40 69XXXXXXX where +49 represents the country code, 40 represents the city code, and the calling party number type is Subscriber. Cisco Unified Communications Manager can be configured with a calling party transformation pattern, along with instructions to strip the country code, city code, and add a prefix of 0. After the instructions are applied, the calling party number displays in the dialed phone as 069XXXXXXX.

Map Globalized Calling Party Number to a Localized Version

To ensure that the phone user does not need to edit the call log directory entry on the phone before placing a call, use route patterns and called party transformation patterns to map the global calling party number to a localized version. This ensures that when the called party returns the call, Cisco Unified Communications Manager can route the call to the correct gateway.

Mapping the global calling party number improves callback functionality so that the called party can return a call without having to modify the directory number in the call log directories on the phone.

Calling Party Normalization Prerequisites

Make sure to activate the Cisco CallManager service in Cisco Unified Serviceability before you configure Calling Party Normalization. For more information, see the Cisco Unified Serviceability Administration Guide.

If you want Cisco Unified Communications Manager to determine the Calling Party Number Type, configure patterns that assign the Calling Party Number Type value that matches the calls that you expect. You can create and apply patterns in the following configuration windows:

• Route patterns
• Hunt pilots
• Translation patterns
• Calling party number transformation patterns

Note

Calling Party Transformation works only with the original calling party. Any modifications done for redirecting numbers affect only the diversion header. Review your configuration from the SIP trunk chapter, and add a diversion header on the SIP trunk itself.
Calling Party Normalization Configuration Task Flow

Calling Party Normalization prefixes and strip digits rules can be applied in a variety of ways in Unified Communications Manager. For example, you can apply digit transformations to device pools, route patterns, translation patterns, hunt pilots, gateways, and trunks. The manner in which you apply digit transformations depends on how you deploy your dial plan, devices, and trunks. For details, review topics relating to dial plans, route patterns, translation patterns, and transformation patterns.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>If you want Unified Communications Manager to determine the calling party number type, create a pattern and configure the Calling Party Number Type that matches the calls that you expect. You can create and apply patterns in the following configuration windows: • Route patterns • Hunt pilots • Translation patterns • Calling party number transformation patterns</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Globalize Calling Party Numbers, on page 585</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Set up Calling Search Spaces, on page 586</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Create Calling Party Transformation Patterns, on page 587</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Apply Calling Party Transformation Patterns to a Calling Search Space, on page 587</td>
</tr>
</tbody>
</table>

Globalize Calling Party Numbers

For incoming calls that arrive via the PSTN, configure settings that will globalize calling party numbers. You can apply settings that will globalize calling party numbers and apply them to a device pool, or to individual devices. Alternatively, you can configure service parameters that will apply calling party normalization settings on a clusterwide basis.

To globalize calling party numbers, perform the following steps:
Procedure

Step 1  If you want to apply calling party normalization settings to particular devices, perform the following steps:
   a) Open the configuration window for the device on which you want to apply settings. For example, device pools, gateways, phones, and trunks.
   b) In the Incoming Calling Party Settings section for the configuration window, apply prefix and strip digit instructions for each calling party number type.

   Note  Cisco Unified Communications Manager includes the prefix in the calling party number field for all additional actions, such as supplementary services including call forwarding, call park, voice messaging, and CDR data that pertain to the call.

Step 2  If you want to use service parameters to globalize calling party numbers on all devices clusterwide, perform the following steps:
   a) From Cisco Unified CM Administration, choose System > Service Parameters.
   b) From the Server drop-down list, select the server on which you want the service to run.
   c) From the Service drop-down list, select Cisco CallManager.
   d) Click Advanced.
   e) Configure values for the following parameters, which can be applied on a clusterwide basis to phones, MGCP gateways, or H.323 gateways:
      • Incoming Calling Party National Number Prefix
      • Incoming Calling Party International Number Prefix
      • Incoming Calling Party Unknown Number Prefix
      • Incoming Calling Party Subscriber Number Prefix

   Note  In order for Cisco Unified Communications Manager to apply the clusterwide service parameter settings on a particular phone, the prefix setting for that phone must be set to the default option at both the device and device pool levels.

Set up Calling Search Spaces

Use this procedure if you are setting up calling search spaces to handle the calling party normalization feature.

Procedure

Step 1  In Cisco Unified CM Administration, choose Call Routing > Class of Control > Partitions.
Step 2  Create partitions for your network.
Step 3  In Cisco Unified CM Administration, choose Call Routing > Class of Control > Calling Search Space.
Step 4  Create calling search spaces for your calling party transformation patterns.
Step 5  For each calling search space, assign partitions to the calling search spaces.
Create Calling Party Transformation Patterns

Use this procedure if you are setting up calling party transformation patterns to handle the calling party normalization feature.

Procedure

Step 1  In Cisco Unified CM Administration, choose Call Routing > Transformation Pattern > Calling Party Transformation Pattern.

Step 2  Create transformation patterns.

Step 3  For each calling party transformation pattern that you create, assign prefixes or strip digits commands that will globalize or localize the calling party number.

Step 4  For each calling party transformation pattern, assign a partition that is associated to one of your calling search spaces.

Apply Calling Party Transformation Patterns to a Calling Search Space

For your devices, assign the incoming Calling Party Transformation CSS to your devices such as device pools, gateways, and trunks.

Procedure

Step 1  In Cisco Unified CM Administration, choose the configuration window that applies to the device on which you want to apply calling party transformations.

  - Gateways
  - Trunks
  - Device Pools

Step 2  To localize calling party numbers, in the Calling Search Space drop-down list box, choose the CSS that contains the calling party transformation pattern that you want to apply.

  Note  If you configure the CSS against the Device Pool, you must also apply that device pool to your phones.

Step 3  To globalize calling party numbers, in the Incoming Calling Party Settings section, choose the calling search space that contains the calling party transformation pattern that you want to apply.

Calling Party Normalization Service Parameter Examples

The following service parameters can be applied on a clusterwide basis to phones, MGCP gateways, or H.323 gateways. In order for a particular device to use the clusterwide parameter, the prefix in the device configuration must be set to Default:.

  - Incoming Calling Party National Number Prefix
• Incoming Calling Party International Number Prefix
• Incoming Calling Party Unknown Number Prefix
• Incoming Calling Party Subscriber Number Prefix

The following table provides examples of prefix and strip digits configurations and how these values can be used to transform the display of the calling party number. For the service parameter configurations, the numbers after the colon represent the number of digits to strip from the beginning of the calling party number while the digits after the colon represent the prefix to add to the beginning of the calling party number.

Table 88: Calling Party Number Normalization Service Parameter Examples

<table>
<thead>
<tr>
<th>Original Calling Number</th>
<th>Service Parameter Value</th>
<th>Description</th>
<th>Final Calling Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>04423452345</td>
<td>+:1</td>
<td>Strip the first digit then add a prefix of +</td>
<td>+4423452345</td>
</tr>
<tr>
<td>04423452345</td>
<td>:2</td>
<td>Strip the first two digits</td>
<td>423452345</td>
</tr>
<tr>
<td>552345</td>
<td>+1:6</td>
<td>Strip the first 6 digits and then add a prefix of +1</td>
<td>+1</td>
</tr>
<tr>
<td>552345</td>
<td>+1:8</td>
<td>Final number is blank because more digits are stripped than are available</td>
<td></td>
</tr>
<tr>
<td>552345</td>
<td>123</td>
<td>Add a prefix of 123</td>
<td>123552345</td>
</tr>
<tr>
<td>blank</td>
<td>+1:2</td>
<td>If calling number is blank no prefix is applied</td>
<td>blank</td>
</tr>
<tr>
<td>0442345</td>
<td>:26</td>
<td>Calling Party Normalization allows only 24 digits to be stripped</td>
<td>Cisco Unified Communications Manager does not allow this configuration</td>
</tr>
</tbody>
</table>

Calling Party Normalization Interactions and Restrictions

Calling Party Normalization Interactions

The following table describes feature interactions with the Calling Party Normalization feature.
## Calling Party Normalization Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transferred Calls</td>
<td>Calling Party Normalization may not be supported for some transferred call scenarios because the transfer feature relies on midcall updates and calling party normalization occurs during initial call setup for each call hop. Following is one example of how calling party normalization can work for transfer. Phone A with extension 12345 and phone number of 972 500 2345 calls Phone B with extension 54321 and phone number 972 500 4321. On Phone B, the calling party number 12345 displays, but Phone B transfers the call through a San Jose gateway to Phone C. During the initial transfer, Phone C displays a calling party number of 972 500 4321, but after the transfer completes, Phone C displays the calling party number for Phone A as 12345.</td>
</tr>
<tr>
<td>Forwarded Calls</td>
<td>Forwarded calls support globalization and localization of calling party numbers. For example, a caller with Phone F calls Phone G in Dallas through the PSTN, but Phone G has forwarded calls to Phone H in San Jose. On the incoming Dallas gateway the calling party number displays as 555-5555/Subscriber, but the call is forwarded to a San Jose gateway. The outgoing call from Dallas displays as 972 555 5555. On the incoming San Jose gateway the +1 is prefixed and Phone F displays a calling number of +1 972 555 5555.</td>
</tr>
<tr>
<td>Call Detail Records</td>
<td>For details of how calling party normalization works with CDR records, see the Cisco Unified Communications Manager Call Detail Records Administration Guide.</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Assistant</td>
<td>Cisco Unified Communications Manager Assistant automatically supports localized and globalized calls if you configure the Calling Party Normalization feature. Cisco Unified Communications Manager Assistant can display localized calling party numbers on the user interfaces. In addition, for an incoming call to the manager, Cisco Unified Communications Manager Assistant can display localized and globalized calling party numbers when filter pattern matching occurs. For information on configuring Cisco Unified Communications Manager Assistant, see the Feature Configuration Guide for Cisco Unified Communications Manager at <a href="http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html">http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html</a>.</td>
</tr>
</tbody>
</table>
InteractionFeature
CiscoUnityConnectiondoesnotsupporttheinternationalescapecharacter (+).Therefore,youmustensurethatcalls toCiscoUnityConnectiondonot containthe,+sothatvoice-messagingfeaturesworkasexpected.


DeviceMobility
TheCallingPartyTransformationCSSoftheroamingdevicepooloverrides thedevice-levelconfigurationofthephoneroamingwithinthesameDevice MobilityGroup,evenwhentheUseDevicePoolCallingPartyTransformation CSScheckboxinthephoneconfigurationwindowremainsunchecked.

Thefollowingexamplesdemonstratehowcallingpartynormalizationworkswithdevice mobilityforaphonewithahomelocationofDallaswhichis currentlyroaminginSanJose.

WhenthephoneisroaminginSanJose,acallcomesthroughthePSTNfrom 9725001212<National>inDallas.OntheincomingSanJosegateway,thecallingpartynumbergetstransformedtotheglobalformatof+14085001212. OnthephonethatcurrentlyisinSanJose,thecallingpartynumberdisplayas 19725001212.

WhenthephoneisroaminginSanJose,acallcomesthroughthePSTNfrom 5001212<Subscriber>fromasevendigitdialingareainSanJose.Onthe incomingSanJosegateway,thecallingpartynumbergetstransformedtothe globalformatof+14085001212.OnthephonethatcurrentlyisinSanJose, thecallingpartynumberdisplayas95001212.

### Calling Party Normalization Restrictions

ThefollowingtabledisplaysrestrictionsthattheCallingPartyNormalizationfeaturehaswithcertainteatures andsystemcomponentsofCiscoUnifiedCommunicationsManager.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unity Connection</td>
<td>Cisco Unity Connection does not support the international escape character (+). Therefore, you must ensure that calls to Cisco Unity Connection do not contain the +, so that voice-messaging features work as expected. For Cisco Unity Connection to work as expected, treat this application as a device and configure calling party transformations that ensure that the + does not get sent to this voice-messaging application. If the Cisco Unity Connection server uses a North American-based dial plan, localize the calling party number to NANP format before Cisco Unity Connection receives the calling party number. Because no calling party transformation options exist in Cisco Unified Communications Manager Administration for voice-messaging ports, make sure that you configure the calling party number transformations in the device pool that is associated with the voice-messaging ports. To localize the calling party number, also consider adding prefixes for access codes so that the voice-messaging application easily can redial the number for certain features, such as Live Reply. For example, you can convert +12225551234 to 912225551234, and you can convert international number, +4423453456, to include the international escape code, 90114423453456.</td>
</tr>
<tr>
<td>Device Mobility</td>
<td>The Calling Party Transformation CSS of the roaming device pool overrides the device-level configuration of the phone roaming within same Device Mobility Group, even when the Use Device Pool Calling Party Transformation CSS check box in the phone configuration window remains unchecked. The following examples demonstrate how calling party normalization works with device mobility for a phone with a home location of Dallas which is currently roaming in San Jose. When the phone is roaming in San Jose, a call comes through the PSTN from 972 500 1212 &lt;National&gt; in Dallas. On the incoming San Jose gateway, the calling party number gets converted to the global format of + 1 408 500 1212. On the phone that currently is in San Jose, the calling party number displays as 1 972 500 1212. When the phone is roaming in San Jose, a call comes through the PSTN from 500 1212 &lt;Subscriber&gt; from a seven-digit dialing area in San Jose. On the incoming San Jose gateway, the calling party number gets converted to the global format of + 1 408 500 1212. On the phone that currently is in San Jose, the calling party number displays as 9 500 1212.</td>
</tr>
</tbody>
</table>
### Table 89: Restrictions with Calling Party Normalization

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Share lines</td>
<td>The calling party number that displays for a shared line depends on the sequence of call control events in Cisco Unified Communications Manager. To avoid displaying an incorrect localized calling party number on a shared line, especially when the shared line occurs in different geographical locations, make sure that you configure the same Calling Party Transformation CSS for different devices that share the same line.</td>
</tr>
<tr>
<td>SIP trunks and MGCP gateways</td>
<td>SIP trunks and MGCP gateways can support sending the international escape character, (+) for calls. H.323 gateways do not support the +. QSIG trunks do not attempt to send the +. For outgoing calls through a gateway that supports +, Cisco Unified Communications Manager can send the + with the dialed digits to the gateway. For outgoing calls through a gateway that does not support +, the international escape character + gets stripped when Cisco Unified Communications Manager sends the call information to the gateway.</td>
</tr>
<tr>
<td>SIP</td>
<td>SIP does not support the number type, so calls through SIP trunks support only the Incoming Number settings for calling party number types of Unknown.</td>
</tr>
<tr>
<td>QSIG</td>
<td>A QSIG configuration usually supports a uniform dial plan. Transformation of numbers and prefixes may cause feature interaction issues if you use QSIG.</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>For localizing the calling party number, the device must apply the transformation by using digit analysis. If you configure the Calling Party Transformation CSS as None, the transformation does not match and does not get applied. Ensure that you configure the Calling Party Transformation Pattern in a non-null partition that is not used for routing.</td>
</tr>
<tr>
<td>T1-CAS and FXO ports</td>
<td>The Calling Party Transformation CSS settings do not apply to T1-CAS and FXO ports on the gateway.</td>
</tr>
<tr>
<td>Cisco Unity Connection</td>
<td>Cisco Unity Connection does not support the international escape character (+). Therefore, you must ensure that calls to Cisco Unity Connection do not contain the +, so that voice-messaging features work as expected. For detailed information on Cisco Unity Connection, go to <a href="http://www.cisco.com/c/en/us/products/unified-communications/unity-connection/index.html">http://www.cisco.com/c/en/us/products/unified-communications/unity-connection/index.html</a>.</td>
</tr>
</tbody>
</table>
Calling Party Normalization Restrictions
Logical Partitioning Overview

With logical partitioning, you can support PSTN and VoIP calls on a single system while meeting regulatory requirements for call separation. For example, under regulatory constraints in India, all calls that are received from or sent to an external phone must be handed off to and carried by a local or long-distance service provider over the full length of the connection, with the applicable toll charges. You can create a single Unified Communications Manager cluster that routes calls appropriately to the PSTN or the VoIP network according to the caller's location and the phone number being called.

Logical partitioning defines which sets of VoIP devices are allowed to communicate with each other. Users do not have to remember to use one line for PSTN and one line for VoIP. Phones making off-net calls are only allowed to talk to a PSTN gateway. It's like having two networks to separately handle your VoIP and PSTN calls, but without the expense of dual infrastructure.

Logical Partitioning Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Enable Logical Partitioning, on page 594</td>
<td>Enable Logical Partitioning.</td>
</tr>
</tbody>
</table>
| Step 2    | To Configure Geolocations, on page 594, perform the following subtasks:  
|           | • Create Geolocations, on page 595  
|           | • Assign Geolocations, on page 595  
|           | • Set the Default Geolocation, on page 596 | Configuring geolocations is a two-step process: defining locations and assigning them to devices. You also can set the default location to be used by all devices in the cluster. |
| Step 3    | Configure a Logical Partitioning Default Policy, on page 596 | Set up a default policy for devices that are not associated with a geolocation or geolocation |
Enable Logical Partitioning

Use this procedure to turn on the Logical Partitioning feature.

**Procedure**

1. **Step 1**
   From Cisco Unified CM Administration, choose **System > Enterprise Parameters**.

2. **Step 2**
   For the **Enable Logical Partitioning** enterprise parameter, choose **True** from the drop-down list.

3. **Step 3**
   Click **Save**.

Configure Geolocations

Configuring geolocations is a two-step process: defining locations and assigning them to devices. You also can set the default location to be used by all devices in the cluster.
### Create Geolocations

Use this procedure to create geolocations that you can assign to the devices in your system. You can use the geolocations for logical partitioning.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Create Geolocations, on page 595</td>
<td>Configure geolocations to specify geographic locations. These are used to associate devices with regulatory features such as logical partitioning. Geolocations are used in policy decisions, such as in-country regulations.</td>
</tr>
<tr>
<td>2</td>
<td>Assign Geolocations, on page 595</td>
<td>Assign a geolocation to a device or device pool.</td>
</tr>
<tr>
<td>3</td>
<td>Set the Default Geolocation, on page 596</td>
<td>Specify a default geolocation for all devices and device pools in this cluster.</td>
</tr>
</tbody>
</table>

#### Create Geolocations

**Procedure**

1. From Cisco Unified CM Administration, choose **System > Geolocation Configuration**.
2. Click **Add New**.
3. Enter a **Name** for the geolocation.
4. Configure the fields on the **Geolocation Configuration** window. For more information on the fields and their configuration options, see the system Online Help.
5. Click **Save**.
6. Repeat this procedure to create additional geolocations.

#### Assign Geolocations

Assign a geolocation to a device or device pool.

**Procedure**

1. From Cisco Unified CM Administration, choose one of the following menu items:
   - **Device > Phone**
   - **Device > Trunk**
   - **Device > Gateway**
   - **System > Device Pool**
2. Perform one of the following tasks:
   - Click **Find** to modify the settings for an existing device or device pool. Enter search criteria, and then choose an existing device or device pool from the resulting list.
• Click Add New to add a new device or device pool. For devices, choose device types and protocols as needed and click Next.

Step 3 From the Geolocation drop-down list, choose a geolocation that you configured.
Step 4 Click Save.

Set the Default Geolocation

Specify a default geolocation for all devices and device pools in this cluster.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Enterprise Parameters.
Step 2 From the Default Geolocation drop-down list, choose a Geolocation that you configured. The default value is Unspecified.
Step 3 Click Save.
Step 4 Click Apply Config.
Step 5 (Optional) If you need to override this default for a specific device or device pool, enter the value on either the Device Configuration or Device Pool Configuration window, and then click Save.

Configure a Logical Partitioning Default Policy

Set up a default policy for devices that are not associated with a geolocation or geolocation filter. The policy allows or denies PSTN calls between these devices.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Logical Partitioning Policy Configuration
Step 2 Click Add New.
Step 3 Configure the fields on the Logical Partition Policy Configuration window. For more information on the fields and their configuration options, see the system Online Help.
Step 4 Click Save.

Note If a policy that contained the value Allow is then later changed to Deny, then it remains Deny. The opposite is also true. A policy previously set to Deny, later changed to Allow is an Allow. The Cisco Unified Reporting > Geolocation Policy Report can help you identify policies that overlap.

Configure Devices to Avoid Logical Partitioning Checks

You can specifically exempt devices and device pools from the partitioning checks.
Configure Geolocation Filters

Logical partitioning assigns a unique identifier to each device based on its location. When one device calls another, these identifiers are used to determine whether the call is allowed and what routing is appropriate. You can choose which fields are used to create this identifier. For example, you can apply different policies based on the room or floor within a building.

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Create Geolocation Filter Rules, on page 597</td>
<td>Geolocation filters allow you to specify which fields are used to create a geolocation identifier. This feature is used to make policy decisions on a subset of the geolocation objects.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign Geolocation Filters, on page 598</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Set the Default Geolocation Filter, on page 598</td>
<td>Configure the Default Geolocation Filter enterprise parameter to specify a default geolocation filter for a cluster. This parameter determines the default geolocation filter setting for all devices and device pools that are not associated with a geolocation filter.</td>
</tr>
</tbody>
</table>

### Create Geolocation Filter Rules

Use this procedure to create geolocation filters that you can use for logical partitioning decisions.
Assign Geolocation Filters

Procedure

**Step 1** From Cisco Unified CM Administration, choose **System > Geolocation Filter**.

**Step 2** Click **Add New**.

**Step 3** Enter a **Name** and **Description** for the filter.

**Step 4** Check the check boxes that correspond to the items you want to use for logical partitioning decisions.

**Step 5** Configure the fields on the **Geolocation Filter Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 6** Click **Save**.

**Step 7** Repeat these steps to create additional geolocation filters.

Set the Default Geolocation Filter

Procedure

**Step 1** From Cisco Unified CM Administration, choose **System > Enterprise Parameters**.

**Step 2** From the **Default Geolocation** drop-down list, choose a Geolocation that you configured. The default value is **Unspecified**.

**Step 3** Click **Save**.

**Step 4** Click **Apply Config**.
Step 5  (Optional) If you need to override this default for a specific device or device pool, specify the default geolocation filter value on either the Device Configuration or Device Pool Configuration window, and then click Save.

Define a Set of Logical Partitioning Policy Records

Define a set of logical partitioning policies for allowing or denying calls between geolocations. Before calls between geolocations are allowed to proceed, the system checks to be sure that calls are allowed between the specified geolocations based on these policies.

Procedure

Step 1  From Cisco Unified CM Administration, choose Call Routing > Logical Partitioning Policy Configuration.
Step 2  Perform one of the following tasks:
   • Click Find to modify the settings for an existing logical partitioning policy. Enter search criteria and then choose an existing logical partitioning policy from the resulting list.
   • Click Add New to add a new logical partitioning policy.
Step 3  Configure the fields on the Logical Partitioning Policy Configuration window. For more information on the fields and their configuration options, see the system Online Help.
   Note  If any policy is left blank without any configuration values, it will become a blank geolocation policy and configuring a Logical Policy for a specific Device Type with the blank Logical Partitioning configurations makes Unified Communications Manager add the policy value (Allow or Deny) in the configured device type.
Step 4  Click Save.

Enable Location Conveyance

Location Conveyance is an optional configuration that lets you share geolocation information across clusters.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Trunk.
Step 2  Do one of the following:
   • Click Find and select an existing trunk.
   • Click Add New to configure a new trunk.
Step 3  Complete the fields in the Trunk Configuration window. For more information on the fields and their configuration options, see the system Online Help.
Step 4  In the Geolocation Information area, select a Geolocation and Geolocation Filter.
Step 5  To enable Location Conveyance, check the Send Geolocation Information check box.
Logical Partitioning Interactions and Restrictions

Logical Partitioning Interactions

Table 90: Logical Partitioning Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
</table>
| Ad Hoc Conference, Join, Join Across Lines, Call Forwarding, Call Transfer | Logical partitioning handling does not take place in the following circumstances:  
  • When all participants are VoIP phones.  
  • When the geolocation or geolocation filter does not associate with a device. |
| Barge, cBarge, and Remote Resume | Logical partitioning handling does not take place in the following circumstances:  
  • When both the caller and the callee devices are VoIP phones, logical partitioning policy checks are ignored.  
  • For the participants in cBarge/Barge, no logical partitioning policy checking exists, and you cannot prevent logical-partitioning-denied scenarios. |
| Cisco Unified Mobility | Logical partitioning handling does not take place in the following circumstances:  
  • Geolocation or geolocation filter does not associate with the involved devices.  
  • No logical partitioning support exists when a dual-mode phone is used. |
| CTI Handling | Logical partitioning handling does not take place in the following circumstances:  
  • When a geolocation or geolocation filter does not associate with any device, handling does not occur.  
  • When all the involved devices specify VoIP phones, handling does not occur. |
| Extension Mobility | Logical partitioning handling does not take place in the following circumstances:  
  • A geolocation or geolocation filter does not associate with a VoIP phone that is logged on to Cisco Extension Mobility, nor does it associate with the calling party or called party device.  
  • The VoIP phone that is logged on to Cisco Extension Mobility calls or receives a call from a VoIP phone. |
Logical Partitioning Restrictions

Table 91: Logical Partitioning Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Barge/cBarge</td>
<td>Barge/cBarge does not occur; the call instance is dropped. For the participants in cBarge/Barge, no logical partitioning policy checking exists, and you cannot prevent logical-partitioning-denied scenarios.</td>
</tr>
<tr>
<td>BLF Presence</td>
<td>BLF Presence notifications are not checked for a logical partitioning policy.</td>
</tr>
<tr>
<td>Cisco Extension Mobility</td>
<td>When Cisco Extension Mobility logs in to a phone in a different geolocation, outgoing PSTN calls can occur when Local Route Groups are configured. Incoming PSTN calls are not placed to the phone but receive a reorder tone.</td>
</tr>
<tr>
<td>Cisco Unified MeetingPlace</td>
<td>The system does not support the logical partitioning feature for calls that involve Cisco Unified MeetingPlace or Cisco Unified MeetingPlace Express.</td>
</tr>
<tr>
<td>Conferences</td>
<td>The logical partitioning checks are not supported for participants across conferences in conference chaining. For example, meet-me and ad hoc chained conferences can have participants that are logical partitioning denied.</td>
</tr>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>H.225 gatekeeper-controlled trunk</td>
<td>Cisco Unified Communications Manager does not communicate geolocation information over a H.225 gatekeeper-controlled trunk.</td>
</tr>
<tr>
<td>H.323 and MGCP Gateways</td>
<td>Cisco Unified Communications Manager does not communicate geolocation info to H.323 or MGCP gateways. Communication to a SIP gateway can be disabled through the SIP trunk check box.</td>
</tr>
<tr>
<td>Mobility Cell Pickup</td>
<td>Logical partitioning deny handling takes place after call is answered on the mobile phone. The logical partitioning policy check does not occur before the call is placed to the mobile phone (as it happens for a basic SNR call). The system checks the logical partitioning policy after the mobile phone answers the call.</td>
</tr>
<tr>
<td>Q.SIG intercluster trunk</td>
<td>Intercluster trunks (ICT) with the Q.SIG protocol are not allowed to communicate geolocation information for the caller or receiving device. The ICT configuration for “Send Geolocation Information” is disabled when the Q.SIG tunneled protocol is selected.</td>
</tr>
<tr>
<td>Reorder Tones</td>
<td>No reorder tone (fast busy tone) is provided on IOS H.323 and SIP gateways upon release of connected calls due to logical partitioning policies.</td>
</tr>
<tr>
<td>Shared Line Active Call</td>
<td>For a restricted logical partitioning scenario, the shared line drops the active call information for the duration of the call, even if a feature moves the shared-line call to the allowed category.</td>
</tr>
<tr>
<td>User Agent Server</td>
<td>The logical partitioning policy checks in the logical partitioning-aware cluster that receives this geolocation may cancel the call if the policy is denied.</td>
</tr>
</tbody>
</table>
### Geolocation and Location Conveyance Overview

Use Geolocations to define the geographical location (or civic address) of devices that is used in policy decisions, such as whether a call from one phone to another is allowed. The Request for Comments (RFC) 4119 standard provides the basis for geolocations.

Use Location Conveyance to allow communication of geolocation information from one cluster to another, when a call is established and during a call.

### Geolocation and Location Conveyance Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | To Configure Geolocations, on page 604, perform the following subtasks:  
  - Configure a Geolocation, on page 604  
  - Assign a Geolocation, on page 604  
  - Set the Default Geolocation, on page 605  
  - Configure Location Conveyance, on page 606 | Configure geolocations to specify geographic locations. These are used to associate devices with regulatory features such as logical partitioning. Geolocations are used in policy decisions, such as in-country regulations. |
| **Step 2** | To Configure Geolocation Filters, on page 606, configure the following subtasks:  
  - Configure a Geolocation Filter, on page 607  
  - Assign a Geolocation Filter, on page 607  
  - Set the Default Geolocation Filter, on page 608 | Configure geolocation filters to choose which fields are used to create a geolocation identifier. This feature is used to make policy decisions on a subset of the geolocation objects. Geolocation filters define which of the geolocation objects should be used when comparing the geolocations of different devices. For example, a group of phones may be... |
Configure Geolocations

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure a Geolocation, on page 604</td>
<td>Configure geolocations to specify geographic locations. These are used to associate devices with regulatory features such as logical partitioning. Geolocations are used in policy decisions, such as in-country regulations.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Assign a Geolocation, on page 604</td>
<td>Assign a geolocation to a device or device pool.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Set the Default Geolocation, on page 605</td>
<td>Specify a default geolocation for all devices and device pools in this cluster.</td>
</tr>
<tr>
<td><strong>Step 4</strong> (Optional) Configure Location Conveyance, on page 606</td>
<td>Configure location conveyance if you want to communicate geolocation information about devices across clusters.</td>
</tr>
</tbody>
</table>

**Configure a Geolocation**

Configure geolocations to specify geographic locations. These are used to associate devices with regulatory features such as logical partitioning. Geolocations are used in policy decisions, such as in-country regulations.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose **System > Geolocation Configuration**.
- **Step 2** Click **Add New**.
- **Step 3** Enter a **Name** for the geolocation.
- **Step 4** Configure the fields on the **Geolocation Configuration** window. For more information on the fields and their configuration options, see the system Online Help.
- **Step 5** Click **Save**.
- **Step 6** Repeat this procedure to create additional geolocations.

**Assign a Geolocation**

Assign a geolocation to a device or device pool.
Procedure

Step 1
From Cisco Unified CM Administration, choose one of the following menu items:

- Device > Phone
- Device > Trunk
- Device > Gateway
- System > Device Pool

Step 2
Perform one of the following tasks:

- Click Find to modify the settings for an existing device or device pool. Enter search criteria, and then choose an existing device or device pool from the resulting list.
- Click Add New to add a new device or device pool. For devices, choose device types and protocols as needed and click Next.

Step 3
From the Geolocation drop-down list, choose a geolocation that you configured.

Step 4
Click Save.

Set the Default Geolocation

Specify a default geolocation for all devices and device pools in this cluster.

Before you begin
Assign a Geolocation, on page 604

Procedure

Step 1
From Cisco Unified CM Administration, choose System > Enterprise Parameters.

Step 2
From the Default Geolocation drop-down list, choose a Geolocation that you configured. The default value is Unspecified.

Step 3
Click Save.

Step 4
Click Apply Config.

Step 5
(Optional) If you need to override this default for a specific device or device pool, enter the value on either the Device Configuration or Device Pool Configuration window, and then click Save.

What to do next

- (Optional) Configure Location Conveyance, on page 606
- Configure Geolocation Filters, on page 606
Configure Location Conveyance

Configure location conveyance if you want to communicate geolocation information about devices across clusters.

Before you begin

- Configure a Geolocation, on page 604  
- Assign a Geolocation, on page 604  
- Set the Default Geolocation, on page 605

Procedure

Step 1  From Cisco Unified CM Administration, choose **Device > Trunk**.

Step 2  Do one of the following:

- Click **Find** and select an existing trunk.
- Click **Add New** to configure a new trunk.

Step 3  Complete the fields in the **Trunk Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 4  In the **Geolocation Information** area, select a **Geolocation** and **Geolocation Filter**.

Step 5  To enable Location Conveyance, check the **Send Geolocation Information** check box.

Step 6  Click **Save**.

Configure Geolocation Filters

Configure geolocation filters to choose which fields are used to create a geolocation identifier. This feature is used to make policy decisions on a subset of the geolocation objects. Geolocation filters define which of the geolocation objects should be used when comparing the geolocations of different devices. For example, a group of phones may be assigned identical geolocations, except for the room and floor in which they are located. Even though the actual geolocations of each phone differ, the filtered geolocation is the same.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure a Geolocation Filter, on page 607</td>
<td>Geolocation filters allow you to specify which fields are used to create a geolocation identifier. This feature is used to make policy decisions on a subset of the geolocation objects.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign a Geolocation Filter, on page 607</td>
<td>Configure the Default Geolocation Filter enterprise parameter to specify a default geolocation filter for a cluster. This parameter</td>
</tr>
<tr>
<td>Step 3</td>
<td>Set the Default Geolocation Filter, on page 608</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td></td>
<td>determines the default geolocation filter setting for all devices and device pools that are not associated with a geolocation filter.</td>
<td></td>
</tr>
</tbody>
</table>

**Configure a Geolocation Filter**

Geolocation filters allow you to specify which fields are used to create a geolocation identifier. This feature is used to make policy decisions on a subset of the geolocation objects.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose System > Geolocation Filter.

**Step 2** Click Add New.

**Step 3** Enter a Name and Description for the filter.

**Step 4** Check the check boxes that correspond to the items you want to use for logical partitioning decisions.

**Step 5** Configure the fields on the Geolocation Filter Configuration window. For more information on the fields and their configuration options, see the system Online Help.

**Step 6** Click Save.

**Step 7** Repeat these steps to create additional geolocation filters.

**Assign a Geolocation Filter**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose one of the following menu items:

- Device > Phone
- Device > Trunk
- Device > Gateway
- System > Device Pool

**Step 2** Perform one of the following tasks:

- Click Find to modify the settings for an existing device or device pool. Enter search criteria and then choose an existing device or device pool from the resulting list.
- Click Add New to add a new device or device pool. For devices, choose device types and protocols as needed and click Next.

**Step 3** From the Geolocation Filter drop-down list, choose a geolocation filter that you configured.

**Step 4** Click Save.
Set the Default Geolocation Filter

Configure the Default Geolocation Filter enterprise parameter to specify a default geolocation filter for a cluster. This parameter determines the default geolocation filter setting for all devices and device pools that are not associated with a geolocation filter.

Before you begin

Assign a Geolocation Filter, on page 607

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Enterprise Parameters.

Step 2  From the Default Geolocation drop-down list, choose a Geolocation that you configured. The default value is Unspecified.

Step 3  Click Save.

Step 4  Click Apply Config.

Step 5  (Optional) If you need to override this default for a specific device or device pool, specify the default geolocation filter value on either the Device Configuration or Device Pool Configuration window, and then click Save.
CHAPTER 82

Configure Location Awareness

• Location Awareness Overview, on page 609
• Location Awareness Prerequisites, on page 611
• Location Awareness Configuration Task Flow, on page 611
• Location Awareness Restrictions, on page 615

Location Awareness Overview

Location Awareness allows administrators to determine the physical location from which a phone connects to the company network. For wireless networks, you can view the wireless access point infrastructure, and which mobile devices currently associate to those access points. For wired networks, you can view the ethernet switch infrastructure and see which devices are currently connect to those switches. This allows you to determine the building, floor, and cube from which a call was placed.

You can view your network infrastructure from the Find and List Switches and Access Points window in Cisco Unified Communications Manager.

This feature updates the Unified Communications Manager database dynamically with the following information:

• Network infrastructure devices such as switches and wireless access points, including IP addresses, hostnames, and BSSID info (where applicable) for each infrastructure device.

• Associated endpoints for each infrastructure device, including:
  • For wireless networks, the list of devices that are currently associated to a wireless access point.
  • For wired networks, the list of devices and device types that are currently connected to an ethernet switch.

Cisco Emergency Responder Integration

Location Awareness helps integrated applications such as Cisco Emergency Responder to determine the physical location of a user who places an emergency call. When Location Awareness is enabled, Cisco Emergency Responder learns of a new device to infrastructure association within minutes of a mobile device associating with a new wireless access point, or a desk phone being connected to a new ethernet switch.

When Cisco Emergency Responder first starts up, it queries the Unified Communications Manager database for the current device to network infrastructure associations. Every two minutes following, Cisco Emergency
Wireless Network Updates

To enable Location Awareness for your wireless infrastructure, you can configure Unified Communications Manager to synchronize with a Cisco Wireless LAN Controller. You can synchronize Unified Communications Manager with up to fifty controllers. During the synchronization process, Unified Communications Manager updates its database with the access point infrastructure that the controller manages. In Cisco Unified CM Administration, you can view the status for your wireless access points, including the list of mobile clients that are associated to each access point.

As mobile clients roam between access points, SIP and SCCP signaling from the endpoint communicates the new device to access point association to Unified Communications Manager, which updates its database. Cisco Emergency Responder also learns of the new association by querying the Unified Communications Manager database every few minutes for new endpoints that have changed their association. As a result, if a mobile client places an emergency call, Cisco Emergency Responder has accurate information on the physical location of the user whom placed the call.

If you have a regular synchronization schedule for your Wireless Access Point controllers, Unified Communications Manager adds and updates access points from the database dynamically following each synchronization.

Using Bulk Administration to insert Access Points

If you are using a third-party wireless access point controller, or if you want to export your access points from Cisco Prime Infrastructure, you can use the Bulk Administration Tool to bulk insert your wireless access point infrastructure from a CSV file into the Unified Communications Manager database. Following the bulk insert, the next location update from the mobile device updates the database with the current access point association.

However, Bulk Administration does not allow you to update your access point infrastructure dynamically as new access points get added to your wireless network. If a mobile call gets placed through an access point that was added after the bulk insert, that access point will not have a record in the database, Unified Communications Manager will not be able to match the BSSID of the new access point, and will mark the infrastructure for the wireless device as UNIDENTIFIED AP.

For detailed information on the Bulk Administration Tool, refer to the "Manage Infrastructure Devices" chapter of the Bulk Administration Guide for Cisco Unified Communications Manager.

Wired Network Updates

No configuration is required to enable Location Awareness for your wired infrastructure—the feature is enabled automatically.

As your wired phones register, signaling between the phone and Cisco Unified Communications Manager updates the database dynamically with the switch infrastructure. You can view details on your company's switch infrastructure in Cisco Unified CM Administration, including the list of phones that are connected to a specific switch.

Unlike mobile devices, wired devices do not typically roam from one switch to another. If a phone does get moved, such as could happen if a worker switches desks within a company, the database gets updated with
the new switch information after the phone re-registers from the new location. In Cisco Unified Communications Manager, the new switch displays the moved phone as a connected endpoint.

If a switch gets deprecated and removed from the network infrastructure, that switch remains visible within Cisco Unified Communications Manager. To remove the old switch from the infrastructure view, you must deactivate the switch from the Access Point and Switch Configuration window.

**Supported Endpoints for Location Awareness**

The following endpoints support tracking via Location Awareness:

- Cisco Unified Wireless IP Phone 7925G
- Cisco Unified Wireless IP Phone 7925G-EX
- Cisco Unified Wireless IP Phone 7926G

These endpoints provide upstream infrastructure information, such as BSSID, through Station Info messages to Cisco Unified Communications Manager. Cisco Emergency Responder uses AXL Change Notifications to track these devices through the associated access point.

For device tracking to work, wireless access points must be defined in Cisco Unified Communications Manager. You can do this by syncing a wireless access point controller or using Bulk Administration to import wireless access point infrastructure.

**Location Awareness Prerequisites**

This feature allows you to synchronize the Cisco Unified Communications Manager database with multiple Cisco Wireless LAN Controllers. You must also set up your Cisco Wireless LAN Controller hardware and your infrastructure of access points. For details, see your controller documentation.

**Location Awareness Configuration Task Flow**

Complete the following tasks to set up Location Awareness in Cisco Unified Communications Manager.

**Before you begin**

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Start Services for Wireless Infrastructure Synchronization, on page 612</td>
<td>In Cisco Unified Serviceability, start services that support the Location Awareness feature.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Wireless Access Point Controller, on page 612</td>
<td>Synchronize the database with a Cisco wireless access point controller. The sync imports the wireless infrastructure into the database. <strong>Tip</strong> Set up a sync schedule for automatic updates.</td>
</tr>
</tbody>
</table>
### Start Services for Wireless Infrastructure Synchronization

Use this procedure to start services that support synchronization with a Cisco Wireless LAN Controller in support of the Location Awareness feature.

**Procedure**

**Step 1**
Log in to Cisco Unified Serviceability and choose **Tools > Service Activation**.

**Step 2**
From the **Server** drop-down list, select the publisher node.

**Step 3**
Make sure that the following services are checked:

- Cisco CallManager
- Cisco AXL Web Service
- Cisco Wireless Controller Synchronization Service

**Step 4**
Optional. If you want to use Bulk Administration to import your network infrastructure from a CSV file, make sure that **Bulk Provisioning Service** is checked.

**Step 5**
Click **Save**.

### Configure Wireless Access Point Controller

Use this procedure to synchronize the database with a Cisco wireless access point controller. During the sync, Unified Communications Manager updates its database with the wireless access point infrastructure that the controller manages. You can add up to fifty wireless access point controllers.
**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose Advanced Features > Device Location Tracking Services > Wireless Access Point Controllers.

**Step 2**  
Select the controller that you want to configure:  
• Click Find and select the controller to edit an existing controller.  
• Click Add New to add a new controller.

**Step 3**  
In the Name field, enter the IP address or hostname for the controller.

**Step 4**  
Enter a Description for the controller.

**Step 5**  
Complete the SNMP settings that will be used for SNMP messaging to the controller:  
   a) From the SNMP Version drop-down list, select the SNMP version protocol that the controller uses.  
   b) Complete the remaining SNMP authentication fields. For more information on the fields and their configuration options, see Online Help.  
   c) Click the Test SNMP Settings to confirm that you entered valid SNMP settings.

**Step 6**  
If you want to configure scheduled syncs to regularly update the database:  
   a) Check the Enable scheduled synchronization to discover Infrastructure Devices check box.  
   b) In the Perform a Re-sync Every fields, create the synchronization schedule.

**Step 7**  
Click Save.

**Step 8**  
(Optional) To update the database immediately, click Synchronize.

**Optional.** If the synchronization pulls in access points that you do not want to track (for example, lab equipment or access points that are not in use) you can remove the access point from tracking.

**Insert Infrastructure Devices**

Use this procedure to complete a bulk import of your wireless access point infrastructure from a CSV file into the Cisco Unified Communications Manager database. You can use this procedure to import a CSV file that was exported from Cisco Prime Infrastructure or if you want to import access points from a third-party wireless access point controller.

**Before you begin**

You must have a data file in comma separated value (CSV) format with the following delineated columns:  
• AccessPoint or Switch Name  
• IPv4 Address  
• IPv6 Address  
• BSSID—Required for Wireless Access Protocol (WAP) infrastructure devices  
• Description—A location identifier, a combination of switch type and location, or another meaningful identifier
You can define both an IPv4 and IPv6 address, or you can define an IPv4 or an IPv6 address.

Note
For the BSSID value, enter the BSSID mask, ending in 0, that uniquely identifies the access point as opposed to the BSSIDs for the individual channels on the access point.

Procedure

Step 1
Choose Bulk Administration > Infrastructure Device > Insert Infrastructure Device. The Insert Infrastructure Device Configuration window displays.

Step 2
In the File Name field, choose the CSV data file that you created for this transaction.

Step 3
In the Job Information area, enter the Job description. The default description is Insert Infrastructure Device.

Step 4
Select when you want to run the job:
- Select the Run Immediately radio button, if you want to run the job immediately.
- Select the Run Later radio button, if you want to schedule the job for later.

Step 5
Click Submit.
If you chose to run the job immediately, the job runs.

Step 6
If you chose to run the job later, schedule when the job runs:
- a) Choose Bulk Administration > Job Scheduler.
- b) Click Find and select the job that you just created.
- c) In the Job Scheduler window, schedule when you want to run the job.
- d) Click Save.
  At the scheduled time, the job runs.

Deactivate Infrastructure Device from Tracking

If the synchronization includes access points or switches that you do not want to track (for example, if the sync pulls in lab equipment or access points that are not in use), you can deactivate the access point or switch from tracking. Unified Communications Manager will not update the status for the access point or switch.

Procedure

Step 1
In Cisco Unified CM Administration, choose Advanced Features > Device Location Tracking Services > Switches and Access Points.

Step 2
Click Find and select the switch or access point that you want to stop tracking.
Step 3  
Click **Deactivate Selected**.

---

**Related Documentation**

After you complete your system configuration, and your system is up and running, you can use tasks in the following chapter to manage your infrastructure on an ongoing basis:

"Manage Infrastructure", *Administration Guide for Cisco Unified Communications Manager and IM and Presence Service*

---

**Location Awareness Restrictions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions and Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Meraki Access Points</td>
<td>The Location Awareness feature does not Meraki access points.</td>
</tr>
</tbody>
</table>
Configure Automated Alternate Routing

- Automated Alternate Routing Overview, on page 617
- AAR Configuration Task Flow, on page 617

Automated Alternate Routing Overview

Configure automated alternate routing (AAR) to automatically reroute calls through the PSTN or other networks when the system blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.

AAR Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Enable Clusterwide Automated Alternate Routing, on page 617</td>
<td>Enable automated alternate routing for the cluster.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure AAR Group, on page 618</td>
<td>Configure automated alternate routing (AAR) to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.</td>
</tr>
</tbody>
</table>

Enable Clusterwide Automated Alternate Routing

Enable Automated Alternate Routing (AAR) for the cluster.

Procedure

- **Step 1**: From Cisco Unified CM Administration, choose **System > Service Parameters**.
- **Step 2**: Select a node in the **Server** drop-down box.
Step 3 From the Service drop-down list, select Cisco Call Manager.

Step 4 In the Clusterwide Parameters (System - CCM Automated Alternate Routing) area, set the Automated Alternate Routing Enable parameter to True.

Configure AAR Group

Configure Automated Alternate Routing (AAR) to automatically reroute calls through the PSTN or other networks when the system blocks a call due to insufficient location bandwidth. With AAR, the caller does not need to hang up and redial the called party.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > AAR Group.

Step 2 Choose one of the following options:

- Click Add New, to add a new AAR group.
- Click Find and choose an AAR group from the resulting list, to modify the settings for an existing AAR group.

The AAR Group Configuration window appears.

Step 3 In the Name field, enter the name that you want to assign to the new AAR group.

The name can contain up to 20 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).

The window refreshes and displays additional fields.

Step 4 Configure the fields on the AAR Group Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 5 Click Save.

Note Optional. To enable AAR to work with hunt pilots, see Hunt Pilot Configuration Task Flow, on page 149.
Configure AS-SIP Endpoints

• AS-SIP Overview, on page 619
• AS-SIP Prerequisites, on page 621
• AS-SIP Endpoint Configuration Task Flow, on page 621

AS-SIP Overview

Assured Services SIP (AS-SIP) endpoints are compliant with MLPP, DSCP, TLS/SRTP, and IPv6 requirements. AS-SIP provides for multiple endpoint interfaces on the Unified Communications Manager.

Many Cisco IP phones support AS-SIP. In addition, the Third-Party AS-SIP Endpoint device type allows a third-party AS-SIP compliant endpoint to be configured and used with Cisco Unified Communications Manager. In addition, the Third-Party AS-SIP Endpoint device type allows a third-party AS-SIP-compliant generic endpoint to be configured and used with Cisco Unified Communications Manager.

AS-SIP Capabilities

The following capabilities are implemented or made available for AS-SIP endpoints:

• MLPP
• TLS
• SRTP
• DSCP for precedence levels
• Error responses
• V.150.1 MER
• Conference Factory flow support
• AS-SIP Line Early Offer

Third-Party AS-SIP Phones

Third-party phones can be provisioned in Cisco Unified Communications Manager using the Third-Party AS-SIP Endpoint device type.
Third-party phones that are running AS-SIP do not get configured through the Cisco Unified Communications Manager TFTP server. The customer must configure them by using the native phone configuration mechanism (usually a web page or TFTP file). The customer must keep the device and line configuration in the Cisco Unified Communications Manager database synchronized with the native phone configuration (for example, extension 1002 on the phone and 1002 in Cisco Unified Communications Manager). Also, if the directory number of a line is changed, the customer must ensure that it gets changed in both Cisco Unified CM Administration and in the native phone configuration mechanism.

**Identification of Third-Party Phones**

The third-party phones that are running SIP do not send a MAC address, they must identify themselves by using username. The REGISTER message includes the following header:

```
Authorization: Digest
username="swhite",realm="ccmsipline",nonce="GBauADss2qW6k9y3hGGVDAqnLfoLk5",uri="sip:172.18.197.224",
algorithm=MD5,response="126c0643a923359ab59d4f53494552e"
```

The username, *swhite*, must match a user that is configured in the **End User Configuration** window of Cisco Unified CM Administration. The administrator configures the SIP third-party phone with the user; for example, *swhite*, in the **Digest User** field of **Phone Configuration** window.

---

**Note**

You can assign each user ID to only one third-party phone. If the same user ID is assigned as the Digest User for multiple phones, the third-party phones to which they are assigned will not successfully register.

---

**Configuration of Third Party AS-SIP Phones and Cisco IP Phones**

The following table provides a comparison overview of the configuration differences between Cisco Unified IP Phones and third-party phones that are running AS-SIP.

**Table 92: Comparison of the Configuration Differences Between Cisco IP Phones and Third-Party Phones**

<table>
<thead>
<tr>
<th>Phone Running AS-SIP</th>
<th>Supports Reset and Restart</th>
<th>Supports Unified Communications Manager Failover and Fallback</th>
<th>Downloads Dial Plan File</th>
<th>Downloads Softkey File</th>
<th>Sends MAC Address</th>
<th>Integrated with Centralized TFTP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IP Phone</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Third-party AS-SIP device</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
</tbody>
</table>

---

**Note**

Not all Cisco IP Phones support AS-SIP. See the phone administration guide for your phone model for support information.
Use Cisco Unified CM Administration to configure third-party phones that are running SIP (see the Configure SIP Profile, on page 324). The administrator must perform configuration steps on the third-party phone that is running SIP; see the following examples:

- Ensure that proxy address in the phone is the IP or Fully Qualified Domain Name (FQDN) of Cisco Unified Communications Manager.
- Ensure directory numbers in the phone match the directory numbers that are configured for the device in Cisco Unified CM Administration.
- Ensure digest user ID (sometimes referred to as Authorization ID) in the phone matches the Digest User ID in the Cisco Unified CM Administration.

For more information, refer to the documentation that came with the third-party phone.

**AS-SIP Conferencing**

MOH is applied to its target (a held party, transferee just before transfer, or conferee just before joining the conference), if the feature invoker (holder, transferor, or conference initiator) supports Cisco-proprietary feature signaling. If the feature invoker does not support Cisco-proprietary feature signaling, then MOH is not applied to its target. Also, if an endpoint explicitly signals that it is a conference mixer, then MOH will not be played to the target. There are two forms of AS-SIP Conferencing:

- Local mixing
- Conference Factory

**Local mixing**

To the Unified CM, the conference initiator simply appears to have established simultaneously active calls, one to each of the other conference attendees. The initiator host the conference locally and the voices are mixed there. The calls from the conference initiator have special signaling that prevent it from being connected to an MOH source.

**Conference Factory**

The conference initiator calls a Conference Factory Server located off a SIP trunk. Through IVR signaling, the conference initiator instructs the Conference Factory to reserve a conference bridge. The Conference Factory gives the numeric address (a routable DN) to the conference initiator, who then establishes a subscription with the bridge to receive conference list information to track the participants. The Conference Factory sends special signaling that prevent it from being connected to an MOH Source.

**AS-SIP Prerequisites**

Determine whether sufficient Device License Units are available. For details, see Cisco Prime License Manager

**AS-SIP Endpoint Configuration Task Flow**

Complete the following tasks to configure an AS-SIP endpoint.
## Configure a Digest User

Use this procedure to configure an end user as a digest user whom uses digest authentication. Devices that are associated to the user will be authenticated via the user's digest credentials.

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure a Digest User, on page 622</td>
<td>Configure the end user to use digest authentication for SIP requests.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure SIP Phone Secure Port, on page 323</td>
<td>Cisco Unified Communications Manager uses this port to listen to SIP phones for SIP line registrations over TLS.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Restart Services, on page 323</td>
<td>After configuring the secure port, restart the Cisco CallManager and Cisco CTL Provider services.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure SIP Profile for AS-SIP, on page 624</td>
<td>Configure a SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks. <strong>Note</strong> The phone-specific parameters are not downloaded to a third-party AS-SIP phone. They are used only by Cisco Unified Communications Manager. Third-party phones must locally configure the same settings.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure Phone Security Profile for AS-SIP, on page 624</td>
<td>You can use the phone security profile to assign security settings such as TLS, SRTP, and digest authentication.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure AS-SIP Endpoint, on page 625</td>
<td>Configure a Cisco IP Phone or a third-party endpoint with AS-SIP support.</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Associate Device with End User, on page 626</td>
<td>Associate the endpoint with a user.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Configure SIP Trunk Security Profile for AS-SIP, on page 626</td>
<td>You can use the sip trunk security profile to assign security features such as TLS or digest authentication to a SIP trunk.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>Configure SIP Trunk for AS-SIP, on page 627</td>
<td>Configure a SIP trunk with AS-SIP support.</td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td>Configure AS-SIP Features, on page 628</td>
<td>Configure additional AS-SIP features such as MLPP, TLS, V.150 and IPv6.</td>
</tr>
</tbody>
</table>
Configure Advanced Call Handling

Configure SIP Phone Secure Port

Follow these steps to configure the SIP Phone Secure Port. Cisco Unified Communications Manager uses this port to listen to SIP phones for SIP line registrations over TLS.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Cisco Unified CM.

Step 2 In the Cisco Unified Communications Manager TCP Port Settings for this Server section, specify a port number in the SIP Phone Secure Port field, or leave the field set to default. The default value is 5061.

Step 3 Click Save.

Step 4 Click Apply Config.

Step 5 Click Ok.

Restart Services

Follow these steps to restart Cisco CallManager and Cisco CTL Provider services.

Procedure

Step 1 From the Cisco Unified Serviceability interface, choose Tools > Control Center - Feature Services.

Step 2 Choose the Cisco Unified Communications Manager server from the Servers drop-down list.

Step 3 Click the radio button that corresponds to the Cisco CallManager service.
Configure SIP Profile for AS-SIP

Use this procedure to configure SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks.

**Procedure**

**Step 1** In Cisco Unified CM Administration, choose **Device > Device Settings > SIP Profile**.

**Step 2** Do either of the following:
- Click **Add New** to create a new SIP Profile.
- Click **Find** and select an existing SIP Profile.

**Step 3** Enter a **Name** and **Description** for the SIP Profile.

**Step 4** Check the **Assured Services SIP conformance** check box.

**Note** This checkbox must be checked for SIP trunks and for third-party AS-SIP phones. It's not mandatory for Cisco IP Phones that support AS-SIP.

**Step 5** In the **Parameters used in Phone** section, configure DSCP precedence values for the types of calls that you expect to make.

**Note** You can also configure DSCP values via clusterwide service parameters. However, the DSCP values within a SIP Profile override the clusterwide settings for all devices that use the SIP Profile.

**Step 6** From the **Early Offer support for voice and video calls** drop-down list, select one of the following options to configure Early Offer support for SIP trunks that use this profile:
- Disabled
- Best Effort (no MTP Inserted)
- Mandatory (insert MTP if needed)

**Step 7** Complete the remaining fields in the **SIP Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

**Step 8** Click **Save**.

Configure Phone Security Profile for AS-SIP

Use this procedure to configure a phone security profile for AS-SIP endpoints. You can use the security profile to assign security settings such as TLS and SRTP.
Configure Advanced Call Handling

Configure AS-SIP Endpoint

Use this procedure to configure an AS-SIP endpoint. Many Cisco IP Phones support AS-SIP. In addition, you can configure AS-SIP for third-party endpoints.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Add New.
Step 3 From the Phone Type drop-down list, select a Cisco IP Phone that supports AS-SIP. Otherwise, select Third-Party AS-SIP Endpoint.
Step 4 Click Next.
Step 5 Configure the following mandatory fields. For more information on the fields and their configuration options, see Online Help.
  - Device Trust Mode—For third-party AS-SIP endpoints only. Select Trusted or Not Trusted.
  - MAC Address
  - Device Pool
Configure Advanced Call Handling

• Phone Button Template
• Owner User ID
• Device Security Profile—Select the phone security profile that you set up for AS-SIP.
• SIP Profile—Select the AS-SIP-enabled SIP Profile that you configured.
• Digest User—Select the user ID that you configure as a digest user. The user must be enabled for digest authentication
• Require DTMF Reception—Check this check box to allow the endpoint to accept DTMF digits.
• Early Offer support for voice and video calls—Check this check box to enable early offer support. This field appears for third-party phones only.

Step 6 Configure the fields in the MLPP and Confidential Access Level Information section.
Step 7 Click Save.
Step 8 Add a Directory Number:
   a) In the left navigation bar, click Add a new DN. The Directory Number Configuration window opens.
   b) Add a Directory Number.
   c) Complete any remaining fields in the Directory Number Configuration window
   d) Click Save.
Step 9 From Related Links, select Configure Device and click Go.
Step 10 Click Apply Config.

Associate Device with End User

Use this procedure to associate an end user to the AS-SIP endpoint.

Procedure

Step 1 From Cisco Unified CM Administration, choose User Management > End User.
Step 2 Click Find and select the user whom you want to associate to the device.
Step 3 In the Device Information section, click Device Association.
The User Device Association window appears.
Step 4 Click Find to view a list of available devices.
Step 5 Select the device that you want to associate, and click Save Selected/Changes.
Step 6 From Related Links, choose Back to User, and click Go.
The End User Configuration window appears, and the associated device that you chose appears in the Controlled Devices pane.

Configure SIP Trunk Security Profile for AS-SIP

Use this procedure to configure a security profile for a SIP trunk that supports AS-SIP
Procedure

Step 1  From Cisco Unified CM Administration, choose System > Security > SIP Trunk Security Profile.
Step 2  Click Add New.
Step 3  Enter a Name for the security profile.
Step 4  From the Device Security Mode drop-down list, choose Authenticated or Encrypted.
Step 5  The Incoming Transport Type and Outgoing Transport Type fields change to TLS automatically.
Step 6  Check the Enable Digest Authentication check box.
Step 7  If you are deploying V.150, configure a value for the SIP V.150 Outbound SDP Offer Filtering drop-down list.
Step 8  Complete the remaining fields in the SIP Trunk Security Profile Configuration window. For more information on the fields and their configuration options, see Online Help.
Step 9  Click Save.

Configure SIP Trunk for AS-SIP

Use this procedure to set up a SIP trunk that supports AS-SIP.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Trunk.
Step 2  Do either of the following:
   • Click Find and select an existing trunk.
   • Click Add New to create a new trunk.
Step 3  For new trunks, from the Trunk Type drop-down list, select SIP Trunk.
Step 4  From the Trunk Service Type drop-down list, select None (Default) and click Next.
Step 5  Enter a Device Name for the trunk.
Step 6  From the Device Pool drop-down list, select a device pool.
Step 7  In the Destination Address field, enter the address of the server to which you are connecting the trunk.
Step 8  From the SIP Trunk Security Profile drop-down list, select the profile that you created for AS-SIP.
Step 9  From the SIP Profile drop-down list, select the SIP Profile that you set up for AS-SIP.
Step 10  Complete any remaining fields in the Trunk Configuration window. For more information on the fields and their configuration options, see Online Help.
Step 11  Click Save.
Configure AS-SIP Features

The procedures in the preceding task flow describe how to configure AS-SIP support on endpoints and trunk. The following table outlines the AS-SIP features that you can deploy and provides configuration reference for each.

<table>
<thead>
<tr>
<th>AS-SIP Feature</th>
<th>Configuration Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Early Offer</td>
<td>SIP Early Offer allows your endpoints to negotiate media during the INVITE request and the 200OK response. There are two modes for Early Offer:</td>
</tr>
<tr>
<td></td>
<td>• Best Effort Early Offer (no MTP Inserted)</td>
</tr>
<tr>
<td></td>
<td>• Mandatory Early Offer (insert MTP if needed)</td>
</tr>
<tr>
<td></td>
<td>Configure Early Offer support via the fields in the following configuration windows. Refer to the online help for detailed field descriptions:</td>
</tr>
<tr>
<td></td>
<td><strong>SIP Profile Configuration</strong> window</td>
</tr>
<tr>
<td></td>
<td>• Early Offer support for voice and video calls—Configure this field to enable Early Offer support on a SIP trunk</td>
</tr>
<tr>
<td></td>
<td>• SDP Session-level Bandwidth Modifier for Early Offer and Re-invite</td>
</tr>
<tr>
<td></td>
<td>• Send send-receive SDP in mid-call INVITE</td>
</tr>
<tr>
<td></td>
<td><strong>Phone Configuration</strong> window (only if the Third Pary AS-SIP Endpoint device type is used)</td>
</tr>
<tr>
<td></td>
<td>• Early Offer support for voice and video calls - check this check box to enable early offer support</td>
</tr>
<tr>
<td>Conference Factory</td>
<td>Specify the URI that an IMS client uses to set up a conference:</td>
</tr>
<tr>
<td></td>
<td>1. From Cisco Unified CM Administration, choose <strong>System &gt; Service Parameters</strong>.</td>
</tr>
<tr>
<td></td>
<td>2. From the <strong>Server</strong> drop-down select your Cisco Unified Communications Manager server.</td>
</tr>
<tr>
<td></td>
<td>3. From the <strong>Service</strong>, select <strong>Cisco CallManager</strong>.</td>
</tr>
<tr>
<td></td>
<td>4. Under <strong>Clusterwide Parameters (Feature - Conference)</strong> assign an IMS Conference Factory URI.</td>
</tr>
<tr>
<td></td>
<td>5. Click <strong>Save</strong>.</td>
</tr>
<tr>
<td><strong>AS-SIP Feature</strong></td>
<td><strong>Configuration Description</strong></td>
</tr>
<tr>
<td>-----------------------------------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>DSCP Markings</td>
<td>DSCP settings allow you to manage QoS and bandwidth within your network. DSCP settings are used to assign a prioritized Traffic Class Label to calls on a per-call basis. You can configure clusterwide DSCP settings via service parameters and you can use the SIP Profile to assign a customized QoS policy for users whom use that profile. For example, you could assign higher priority for the calls of an executive (for example, a CEO) or a sales team to ensure that their calls are not dropped if network bandwidth issues arise. To configure DSCP, see DSCP Settings Configuration Task Flow, on page 483.</td>
</tr>
<tr>
<td>IPv6</td>
<td>By default, Cisco Unified Communications Manager is configured to use IPv4 addressing. However, you can configure the system to support the IPv6 stack thereby allowing you to deploy a SIP network with IPv6-only endpoints. To configure IPv6, see Dual Stack IPv6 Configuration Task Flow, on page 90.</td>
</tr>
<tr>
<td>Multilevel Precedence and Preemption (MLPP)</td>
<td>The Multilevel Precedence and Preemption (MLPP) service allows placement of priority calls. This capability assures high-ranking personnel of communication to critical organizations and personnel during network stress situations, such as a national emergency or degraded network situations. To configure MLPP, see Multilevel Precendence and Preemption Task Flow, on page 631.</td>
</tr>
<tr>
<td>Secure Real-Time Transport Protocol (SRTP)</td>
<td>The Secure Real-time Transport Protocol (SRTP) can be used to provide encryption and authentication to media streams in your calls. SRTP can be configured for phones within the Phone Security Profile Configuration that the phone uses. You must set the Device Security Mode field to Encrypted.</td>
</tr>
</tbody>
</table>
## Configuration Description

The V.150 Minimum Essential Requirements feature allows you to make secure calls in a modem over IP network. The feature uses a dialup modem for large installed bases of modems and telephony devices operating on a traditional public switched telephone network (PSTN).


<table>
<thead>
<tr>
<th>AS-SIP Feature</th>
<th>Configuration Description</th>
</tr>
</thead>
</table>
Multilevel Precedence and Preemption Overview

The Multilevel Precedence and Preemption (MLPP) service allows placement of priority calls. Properly validated users can preempt lower priority phone calls with higher priority calls. An authenticated user can preempt calls either to targeted stations or through fully subscribed TDM trunks. This capability assures high-ranking personnel of communication to critical organizations and personnel during network stress situations, such as a national emergency or degraded network situations.

Multilevel Precedence and Preemption Prerequisites

Supported SCCP or SIP phones. See the Cisco IP Phone Administration Guide and Cisco IP Phone User Guide for your phones for feature support and more information.

Multilevel Precendence and Preemption Task Flow

Before you begin

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>To Configure Domains and Domain Lists, on page 633, perform the following subtasks:</td>
<td>Configure an MLPP domain to specify the devices and resources that are associated with an MLPP subscriber.</td>
</tr>
<tr>
<td></td>
<td>• Configure a Multilevel Precedence and Preemption Domain, on page 634</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure a Resource Priority Namespace Network Domain, on page 634</td>
<td></td>
</tr>
</tbody>
</table>
### Configure Advanced Call Handling

#### Multilevel Precedence and Preemption Task Flow

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Configure a Resource Priority Namespace Network Domain List, on page 635</td>
<td>A common device configuration includes MLPP-related information that can be applied to multiple users and their devices. Ensure that each device is associated with a common device configuration. These settings override the enterprise parameter settings.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure a Common Device Configuration for Multilevel Precedence and Preemption, on page 635</td>
<td>Set enterprise parameters to enable MLPP indication and preemption. If individual devices and devices in common device configurations have MLPP settings of Default, the MLLP-related enterprise parameters apply to these devices and common device configurations.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure the Enterprise Parameters for Multilevel Precedence and Preemption, on page 636</td>
<td>Configure a partition to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Devices that are typically placed in partitions include DNs and route patterns. These entities associate with DNs that users dial. For simplicity, partition names usually reflect their characteristics.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure a Partition for Multilevel Precedence and Preemption, on page 637</td>
<td>Configure a calling search space is an ordered list of partitions. Calling search spaces determine the partitions that calling devices, including IP phones, softphones, and gateways, can search when attempting to complete a call.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure a Calling Search Space for Multilevel Precedence and Preemption, on page 638</td>
<td>Configure route patterns to route or block both internal and external calls.</td>
</tr>
<tr>
<td><strong>Step 6</strong> Configure a Route Pattern for Multilevel Precedence and Preemption, on page 639</td>
<td>Configure translation patterns to specify how to route a call after it is placed. Configuring translation patterns allows your system to manipulate calling and called digits as needed. When the system identifies that a pattern match occurred, your system uses the calling search space that is configured for the translation pattern to perform the subsequent match.</td>
</tr>
<tr>
<td><strong>Step 7</strong> Configure a Translation Pattern for Multilevel Precedence and Preemption, on page 640</td>
<td>Configure Cisco Unified Communications Manager to communicate with non-IP telecommunications devices.</td>
</tr>
<tr>
<td><strong>Step 8</strong> Configure Multilevel Precedence and Preemption for Gateways, on page 641</td>
<td><strong>Step 9</strong> Configure Multilevel Precedence and Preemption for Phones, on page 642</td>
</tr>
</tbody>
</table>
Configure Advanced Call Handling

Configure Domains and Domain Lists

Configure an MLPP domain to specify the devices and resources that are associated with an MLPP subscriber.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure a Multilevel Precedence and Preemption Domain, on page 634</td>
</tr>
</tbody>
</table>

Associate devices and resources with an MLPP subscriber. When an MLPP subscriber that belongs to a particular domain places a precedence call to another MLPP subscriber that belongs to the same domain, the MLPP service can preempt the existing call that the called MLPP subscriber is on for a higher precedence call. MLPP service availability does not span across different domains.

The MLPP domain subscription of the originating user determines the domain of the call and its connections. Only higher precedence calls in one domain can preempt connections that calls in the same domain are using.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure a Resource Priority Namespace Network Domain, on page 634</td>
</tr>
</tbody>
</table>

Configure namespace domains for a Voice over Secured IP (VoSIP) network that uses SIP trunks. Your system prioritizes the SIP-signaled...
### Configure a Resource Priority Namespace Network Domain

Configure namespaces for a Voice over Secured IP (VoSIP) network that uses SIP trunks. Your system prioritizes the SIP-signaled resources so that those resources can be used most effectively during emergencies and congestion of telephone circuits, IP bandwidth, and gateways. Endpoints receive the precedence and preemption information.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 3 Configure a Resource Priority Namespace Network Domain List, on page 635</td>
<td>Configure a list of acceptable network domains. Incoming calls are compared to the list and processed, if an acceptable network domain is in the list.</td>
</tr>
</tbody>
</table>

### Configure a Multilevel Precedence and Preemption Domain

Associate devices and resources with an MLPP subscriber. When an MLPP subscriber that belongs to a particular domain places a precedence call to another MLPP subscriber that belongs to the same domain, the MLPP service can preempt the existing call that the called MLPP subscriber is on for a higher precedence call. MLPP service availability does not span across different domains.

The MLPP domain subscription of the originating user determines the domain of the call and its connections. Only higher precedence calls in one domain can preempt connections that calls in the same domain are using.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose <strong>System &gt; MLPP &gt; Domain &gt; MLPP Domain</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Add New</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the <strong>Domain Name</strong> field, enter the name that you want to assign to the new MLPP domain. You can enter up to 50 alphanumeric characters, and any combination of spaces, periods (.), hyphens (-), and underscore characters (_).</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Domain ID</strong> field, enter a unique six-character hexadecimal MLPP domain ID. Domain IDs must fall in the range between 000001 and FFFFFF. (000000 is reserved for the default MLPP domain ID.)</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>
Configure Advanced Call Handling

Procedure

Step 1 From Cisco Unified CM Administration, System > MLPP > Namespace > Resource Priority Namespace Network Domain.

Step 2 Enter the name for the Resource Priority Namespace Network Domain in the information section. The maximum number of domain names is 100.

Step 3 Enter a description for the domain name.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), or angle brackets (<>).

Step 4 Check the Make this the Default Resource Priority Namespace Network Domain check box if you want the domain name to be the default.

Step 5 Click Save.

Configure a Resource Priority Namespace Network Domain List

Configure a list of acceptable network domains. Incoming calls are compared to the list and processed, if an acceptable network domain is in the list.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > MLPP > Namespace > Resource Priority Namespace List.

Step 2 Enter the name for the Resource Priority Namespace List. The maximum number of characters is 50.

Step 3 Enter a description for the list. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

Step 4 Use the Up and Down Arrows to move a Resource Priority Namespace Network Domain to the Selected Resource Priority Namespaces field.

Step 5 Click Save.

Configure a Common Device Configuration for Multilevel Precedence and Preemption

A common device configuration includes MLPP-related information that can be applied to multiple users and their devices. Ensure that each device is associated with a common device configuration. These settings override the enterprise parameter settings.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Common Device Configuration.

Step 2 Perform one of the following tasks:
Configure the Enterprise Parameters for Multilevel Precedence and Preemption

Set enterprise parameters to enable MLPP indication and preemption. If individual devices and devices in common device configurations have MLPP settings of Default, the MLLP-related enterprise parameters apply to these devices and common device configurations.

Procedure

Step 1 Choose System > Enterprise Parameters.
Step 2 Configure the MLPP enterprise parameters on the Enterprise Parameters Configuration window. See the Related Topics section for more information about the parameters and their configuration options.
Step 3 Click Save.

Enterprise Parameters for Multilevel Precedence and Preemption

Table 93: Enterprise Parameters for Multilevel Precedence and Preemption

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain Identifier</td>
<td>Set this parameter to define a domain. Because MLPP service applies to a domain, Cisco Unified Communications Manager marks only connections and resources that belong to calls from MLPP users in a given domain with a precedence level. Cisco Unified Communications Manager can preempt only lower precedence calls from MLPP users in the same domain. The default is 000000.</td>
</tr>
<tr>
<td>MLPP Indication Status</td>
<td>This parameter specifies whether devices use MLPP tones and special displays to indicate MLPP precedence calls. To enable MLPP indication across the enterprise, set this parameter to MLPP Indication turned on. The default is MLPP Indication turned off.</td>
</tr>
<tr>
<td>MLPP Preemption Setting</td>
<td>This parameter determines whether devices should apply preemption and preemption signaling (such as preemption tones) to accommodate higher precedence calls. To enable MLPP preemption across the enterprise, set this parameter to Forceful Preemption. The default is No preemption allowed.</td>
</tr>
</tbody>
</table>
In a precedence call, if the called party subscribes to alternate party diversion, this timer indicates the seconds after which Cisco Unified Communications Manager will divert the call to the alternate party if the called party does not acknowledge preemption or does not answer a precedence call. The default is 30 seconds.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Precedence Alternate Party Timeout</td>
<td>This parameter determines whether a precedence call will forward to the voice-messaging system. If the parameter is set to False, precedence calls do not forward to the voice-messaging system. If the parameter is set to True, precedence calls forward to the voice-messaging system. For MLPP, the recommended setting for this parameter is False, as users, not the voice-messaging system, should always answer precedence calls. The default is False.</td>
</tr>
</tbody>
</table>

**Configure a Partition for Multilevel Precedence and Preemption**

Configure a partition to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Devices that are typically placed in partitions include DNs and route patterns. These entities associate with DNs that users dial. For simplicity, partition names usually reflect their characteristics.

**Procedure**

1. From Cisco Unified CM Administration, choose **Call Routing > Class of Control > Partition**.
2. Click **Add New** to create a new partition.
3. In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan. Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.
4. Enter a comma (,) after the partition name and enter a description of the partition on the same line. The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]). If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.
5. To create multiple partitions, use one line for each partition entry.
6. From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition. The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.
7. Select one of the following radio buttons to configure the **Time Zone**:
Partition Naming Guidelines

The list of partitions in a calling search space is limited to a maximum of 1024 characters. This means that the maximum number of partitions in a CSS varies depending on the length of the partition names. Use the following table to determine the maximum number of partitions that you can add to a calling search space if partition names are of fixed length.

<table>
<thead>
<tr>
<th>Partition Name Length</th>
<th>Maximum Number of Partitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>2 characters</td>
<td>340</td>
</tr>
<tr>
<td>3 characters</td>
<td>256</td>
</tr>
<tr>
<td>4 characters</td>
<td>204</td>
</tr>
<tr>
<td>5 characters</td>
<td>172</td>
</tr>
<tr>
<td>...</td>
<td>...</td>
</tr>
<tr>
<td>10 characters</td>
<td>92</td>
</tr>
<tr>
<td>15 characters</td>
<td>64</td>
</tr>
</tbody>
</table>

Configure a Calling Search Space for Multilevel Precedence and Preemption

A calling search space is an ordered list of partitions. Calling search spaces determine the partitions that calling devices, including IP phones, softphones, and gateways, can search when attempting to complete a call.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Calling Search Space.
Step 2 Click Add New.
Step 3 In the Name field, enter a name.
   Ensure that each calling search space name is unique to the system. The name can include up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).
Step 4 In the Description field, enter a description.
Configure a Route Pattern for Multilevel Precedence and Preemption

Configure route patterns to route or block both internal and external calls.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Route Pattern.
Step 2 Perform one of the following tasks:
   • To modify the settings for an existing route pattern, enter search criteria, click Find, and then choose an existing route pattern from the resulting list.
   • To add a new route pattern, click Add New.
Step 3 Configure the fields on the Route Pattern Configuration window. See the Related Topics section for more information about the fields and their configuration options.
Step 4 Click Save.

Route Pattern Configuration Fields for Multilevel Precedence and Preemption

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>Enter the route pattern, including numbers and wildcards, without spaces. For example, for NANP, enter 9.@ for typical local access or 8XXX for a typical private network numbering plan. Valid characters include the uppercase characters A, B, C, and D and +, which represents the international escape character +.</td>
</tr>
</tbody>
</table>
Configure a Translation Pattern for Multilevel Precedence and Preemption

Configure translation patterns to specify how to route a call after it is placed. Configuring translation patterns allows your system to manipulate calling and called digits as needed. When the system identifies that a pattern match occurred, your system uses the calling search space that is configured for the translation pattern to perform the subsequent match.

Procedure

Step 1  In Cisco Unified CM Administration, choose Call Routing > Translation Pattern.

Step 2  Perform one of the following tasks:

Field | Description
--- | ---
MLPP Precedence | Choose an MLPP precedence setting for this route pattern from the drop-down list:
  - Executive Override—Highest precedence setting for MLPP calls.
  - Flash Override—Second highest precedence setting for MLPP calls.
  - Flash—Third highest precedence setting for MLPP calls.
  - Immediate—Fourth highest precedence setting for MLPP calls.
  - Priority—Fifth highest precedence setting for MLPP calls.
  - Routine—Lowest precedence setting for MLPP calls.
  - Default—Does not override the incoming precedence level but rather lets it pass unchanged.

Apply Call Blocking Percentage | Check this check box to enable the Destination Code Control (DCC) feature. By enabling DCC, all calls other than flash and higher precedence calls made to the destination are filtered and allowed or disallowed based on the Call Blocking Percentage quota set for the destination. Flash and higher precedence calls are allowed at all times. DCC is disabled by default.

The Apply Call Blocking Percentage field is enabled only if the MLPP level is immediate, priority, routine or default.

Call Blocking Percentage (%) | Enter the percentage of calls to be blocked for this destination in numerals. This value specifies the percentage of lower precedence calls made to this destination that get blocked by the route pattern. This percentage limits the lower precedence calls only; the flash and higher precedence calls made to this destination are allowed at all times.

The Call Blocking Percentage (%) field is enabled only if the Apply Call Blocking Percentage check box is checked.

Configure Multilevel Precedence and Preemption for Gateways

Configure Cisco Unified Communications Manager to communicate with non-IP telecommunications devices.

**Before you begin**

- Configure one of the following gateways:
  - Cisco Catalyst 6000 24 port FXS Gateway
  - Cisco Catalyst 6000 E1 VoIP Gateway
  - Cisco Catalyst 6000 T1 VoIP Gateway
  - Cisco DE-30+ Gateway
  - Cisco DT-24+ Gateway
  - H.323 Gateway

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Device > Gateway

**Step 2**
Perform one of the following tasks:

- To modify the settings for an existing gateway, enter search criteria, click **Find**, and choose a gateway from the resulting list.
- To add a new gateway:
  a. Click **Add New**.
b. From the Gateway Type drop-down list, choose one of the supported gateway models.

c. Click Next.

**Step 3** Configure the MLPP fields on the **Gateway Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

**Step 4** Click **Save**.

---

## Configure Multilevel Precedence and Preemption for Phones

**Caution** 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.

### Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone**.

**Step 2** Enter search criteria.

**Step 3** Click **Find** and choose a phone from the resulting list.

**Step 4** Configure the MLPP fields on the **Phone Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

---

## Multilevel Precedence and Preemption Settings for Phones

**Table 96: Multilevel Precedence and Preemption Settings for Phones**

<table>
<thead>
<tr>
<th>MLPP Settings for Phones</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Field</strong></td>
<td></td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>Choose the common device configuration that you configured. The common device configuration includes the attributes (services or features) that are associated with a particular user.</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>From the drop-down list, choose a calling search space (CSS) that you configured. A calling search space comprises a collection of partitions that are searched to determine how a dialed number should be routed. The calling search space for the device and the calling search space for the directory number are used together. The directory number CSS takes precedence over the device CSS.</td>
</tr>
</tbody>
</table>
### MLPP Settings for Phones

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
<td>Choose an MLPP domain from the drop-down list for the MLPP domain that is associated with this device. If you leave the None value, this device inherits its MLPP domain from the value that was set in the common device configuration. If the common device configuration 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.</td>
</tr>
</tbody>
</table>
| MLPP Indication   | If available, this setting specifies whether a device that can play precedence tones will use the capability when it places an MLPP precedence call. From the drop-down list, choose a setting to assign to this device from the following options:  
  - Default — This device inherits its MLPP indication setting from the common device configuration.  
  - Off — This device does not handle nor process indication of an MLPP precedence call.  
  - On — This device handles and processes 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.  
  Turning on MLPP Indication (at the enterprise parameter or device level) disables normal Ring Setting behavior for the lines on a device, unless MLPP Indication is turned off (overridden) for the device. |
| MLPP Preemption   | Be aware that this setting is not available on all devices. If available, this setting specifies whether a device that can preempt calls in progress will use the capability when it places an MLPP precedence call. From the drop-down list, choose a setting to assign to this device from the following options:  
  - Default  
    — This device inherits its MLPP preemption setting from the common device configuration.  
  - 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. |
Configure a Directory Number to Place Multilevel Precedence and Preemption Calls

After you configure a device, you can add a line (directory number) from the updated Device Configuration window.

Procedure

**Step 1**
From Cisco Unified CM Administration in the Device Configuration window, click Add a new DN for the appropriate line.

**Step 2**
In the Target (Destination) field, enter the number to which MLPP precedence calls should be diverted if this directory number receives a precedence call and neither this number nor its call forward destination answers the precedence call.

Values can include numeric characters, octothorpe (#), and asterisk (*).

**Step 3**
From the MLPP Calling Search Space drop-down list, choose the calling search space to associate with the MLPP alternate party target (destination) number.

**Step 4**
In the MLPP No Answer Ring Duration (seconds), enter the number of seconds (between 4 and 60) after which an MLPP precedence call is directed to this directory number alternate party if this directory number and its call-forwarding destination have not answered the precedence call.

Leave this setting blank to use the value that is set in the Precedence Alternate Party Timeout enterprise parameter.

**Step 5**
Click Save.

Configure a User Device Profile for Multilevel Precedence and Preemption

When a user profile is assigned to a phone, the phone inherits the configuration of the assigned user, including any CSS that is associated with the user. The phone CSS can, however, override the user profile. Cisco Unified Communications Manager assigns the precedence level that is associated with the dialed pattern to the call when a pattern match occurs. The system sets the call request as a precedence call with the assigned precedence level.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose Device > Device Settings > Device Profile.

**Step 2**
Perform one of the following tasks:

- To modify the settings for an existing device profile, enter search criteria, click Find, and then choose an existing device profile from the resulting list.
- To add a new device profile:
  - Click Add New.
  - From the Device Profile Type drop-down list, choose a profile type.
Configure Advanced Call Handling

Configure the Default Device Profile for Multilevel Precedence and Preemption

- Click **Next**.
- From the **Device Protocol** drop-down list, choose either **SIP** or **SCCP**.

**Step 3**  Click **Next**.
**Step 4**  From the **MLPP Domain** drop-down list, choose an MLLP domain that you configured.
**Step 5**  From the **MLPP Indication** drop-down list, choose one of the following settings to specify whether a device that is capable of playing precedence tones will use the capability when it places an MLPP precedence call:
  - **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.

**Step 6**  From the **MLPP Preemption** drop-down list, choose one of the following settings to specify whether a device that is capable of preempting calls in progress will use the capability when it places an MLPP precedence call:
  - **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.

**Step 7**  Click **Save**.

---

**Configure the Default Device Profile for Multilevel Precedence and Preemption**

Use the default device profile for whenever a user logs on to a phone model for which no user device profile exists. A default device profile comprises the set of services and features that are associated with a particular device.

⚠️ **Caution**

Do not configure a default device profile 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.

---

**Procedure**

**Step 1**  In Cisco Unified CM Administration, choose Device > Device Settings > Default Device Profile.

**Step 2**  Perform one of the following tasks:
  - To modify the settings for an existing default device profile, choose an existing default device profile from the Device Profile Defaults section.
To add a new default device profile, choose a device profile type from the drop-down list, click Next, choose a device protocol, and then click Next.

**Step 3**  
From the **MLPP Domain** drop-down list, choose an MLPP domain that you configured to associate to the device.

**Step 4**  
From the **MLPP Indication** drop-down list, choose one of the following settings to specify whether a device that is capable of playing precedence tones will use the capability when it places an MLPP precedence call:

- **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.

**Step 5**  
From the **MLPP Preemption** drop-down list, choose one of the following settings to specify whether a device that is capable of preemption calls in progress will use the capability when it places an MLPP precedence call:

- **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.

**Step 6**  
Click Save.

---

### Multilevel Precedence and Preemption Interactions and Restrictions

#### Multilevel Precedence and Preemption Interactions

Table 97: Multilevel Precedence and Preemption Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>729 Annex A</td>
<td>729 Annex A is supported.</td>
</tr>
<tr>
<td>Cisco Extension Mobility</td>
<td>The MLPP service domain remains associated with a user device profile when a user logs in to a device by using extension mobility. The MLPP Indication and Preemption settings also propagate with extension mobility. If either the device or the device profile do not support MLPP, these settings do not propagate.</td>
</tr>
</tbody>
</table>
Interaction with Cisco Unified Communications Manager Assistant as follows:

- When Cisco Unified Communications Manager Assistant handles an MLPP precedence call, Cisco Unified Communications Manager Assistant preserves call precedence.
- Cisco Unified Communications Manager Assistant filters MLPP precedence calls in the same manner as it filters all other calls. The precedence of a call does not affect whether the call is filtered.
- Because Cisco Unified Communications Manager Assistant does not register the precedence of a call, it does not provide any additional indication of the precedence of a call on the assistant console.

Immediate Divert diverts calls to voice-messaging mailboxes regardless of the type of call (for example, a precedence call). When Alternate Party Diversion (call precedence) is activated, Call Forward No Answer (CFNA) is also deactivated.

RSVP supports MLPP inherently. The Cisco Unified Communications Manager System Guide explains how MLPP functions when RSVP is activated.

MLPP interacts with multiple line appearances, call transfer, call forwarding, three-way calling, call pickup, and hunt pilots as documented in the and the subsections that describe the interaction with each service.

### Multilevel Precedence and Preemption Restrictions

Table 98: Multilevel Precedence and Preemption Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bandwidth</td>
<td>Cisco Unified Communications Manager preempts lower precedence calls when adjusting video bandwidth for high priority calls. If the bandwidth is not sufficient to preempt, Cisco Unified Communications Manager instructs endpoints to use previously reserved lower video bandwidth. When Cisco Unified Communications Manager preempts a video call, the preempted party receives a preemption tone and the call gets cleared.</td>
</tr>
<tr>
<td>Call Detail Records</td>
<td>For the DRSN, CDRs represent precedence levels with values 0, 1, 2, 3, and 4 where 0 specifies Executive Override and 4 specifies Routine, as used in DSN. CDRs thus do not use the DRSN format.</td>
</tr>
<tr>
<td>Common Network Facility Preemption</td>
<td>Common Network Facility Preemption support exists only for T1-CAS and T1-PRI (North American) interfaces on targeted Voice over IP gateways that Cisco Unified Communications Manager controls by using MGCP protocol and that have been configured as MLPP Preemption Enabled.</td>
</tr>
</tbody>
</table>
## Multilevel Precedence and Preemption Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercluster trunks</td>
<td>Intercluster trunk MLPP carries precedence information through dialed digits. Domain information does not get preserved and must be configured per trunk for incoming calls.</td>
</tr>
<tr>
<td>Line Groups</td>
<td>MLPP-enabled devices are not supported in line groups. We recommend the following guidelines:</td>
</tr>
<tr>
<td></td>
<td>• MLPP-enabled devices should not be configured in a line group. Route groups, however, are supported. Both trunk selection and hunting methods are supported.</td>
</tr>
<tr>
<td></td>
<td>• If an MLPP-enabled device is configured in a line group or route group, in the event of preemption, if the route list does not lock onto the device, the preempted call may be rerouted to other devices in the route/hunt list and preemption indication may be returned only after no devices are able to receive the call.</td>
</tr>
<tr>
<td></td>
<td>• Route lists can be configured to support either of two algorithms of trunk selection and hunting for precedence calls. In method 1, perform a preemptive search directly. In method 2, first perform a friendly search. If this search is not successful, perform a preemptive search. Method 2 requires two iterations through devices in a route list. If route lists are configured for method 2, in certain scenarios involving line groups, route lists may seem to iterate through the devices twice for precedence calls.</td>
</tr>
<tr>
<td>Look Ahead For Busy</td>
<td>Cisco Unified Communications Manager does not support the Look Ahead for Busy (LFB) option.</td>
</tr>
<tr>
<td>MLPP Notification</td>
<td>Only MLPP Indication Enabled devices generate MLPP-related notifications, such as tones and ringers. If a precedence call terminates at a device that is not MLPP Indication Enabled, no precedence ringer gets applied. If a precedence call originates from a device that is not MLPP Indication Enabled, no precedence ringback tone gets applied. If a device that is not MLPP Indication Enabled is involved in a call that is preempted (that is, the other side of the call initiated preemption), no preemption tone gets applied to the device.</td>
</tr>
<tr>
<td>Phones and trunks</td>
<td>For phones, devices that are MLPP indication disabled (that is, MLPP Indication is set to Off) cannot be preempted. For trunks, MLPP indication and preemption function independently.</td>
</tr>
<tr>
<td>Ring Setting Behavior</td>
<td>Turning on MLPP Indication (at the enterprise parameter, common device configuration, or device level) disables normal Ring Setting behavior for the lines on a device, unless MLPP Indication is turned off (overridden) for the device.</td>
</tr>
<tr>
<td>SCCP</td>
<td>IOS gateways support the SCCP interface to Cisco Unified Communications Manager. They support BRI and analog phones which appear on Cisco Unified Communications Manager as supported phone models. SCCP phones support the MLPP feature, and so do some phones with specific SIP loads. See the relevant phone administration and user guides for Cisco IP phone support information.</td>
</tr>
</tbody>
</table>
Supplementary Services
MLPP support for supplementary services specifies the following restrictions:

- MLPP addresses only the basic Call Pickup feature and Group Call Pickup feature, not Other Group Pickup.
- Call Forward All (CFA) support for inbound MLPP calls always forwards the call to the MLPP Alternate Party (MAP) target of the called party, if the MAP target is configured. In the event of an incorrect configuration (that is, if no MAP target is specified), the call is rejected, and the calling party receives a reorder tone.
- Call Forward No Answer (CFNA) support for inbound MLPP calls forwards the call once to a CFNA target. After the first hop, if the call is unanswered, the call is sent to the MAP target of the original called party, if the MAP target has been configured. In the event of an incorrect configuration (that is, if no MAP target is specified), the call gets rejected, and the calling party receives reorder tone.
- Call Forward Busy (CFB) support for inbound MLPP calls forwards the call up to the maximum number that has been configured for forwarding hops. If the maximum hop count gets reached, the call gets sent to the MAP target of the original called party, if the MAP target has been configured. In the event of an incorrect configuration (that is, no MAP target is specified), the call gets rejected, and the calling party receives reorder tone.
- For hunt pilot support, the hunt group algorithm must specify Longest Idle Time, Top Down, or Circular. Ensure the hunt group options for busy treatment, no answer treatment, and unregistered treatment are set to Try next member, but do not go to next group. Preemption only occurs across a single hunt group.

User Access Channel
User Access Channel support exists only for the following Cisco Unified IP Phone models, which must be configured as MLPP Preemption Enabled:

- Cisco Unified IP Phone 7960, 7962, 7965
- Cisco Unified IP Phone 7940, 7942, 7945
Multilevel Precedence and Preemption Restrictions
PART XI

Reference Information

• Cisco Unified Communications Manager TCP and UDP Port Usage, on page 653
• Port Usage Information for the IM and Presence Service, on page 671
Cisco Unified Communications Manager TCP and UDP Port Usage

This chapter provides a list of the TCP and UDP ports that Cisco Unified Communications Manager uses for intracluster connections and for communication with external applications or devices. You will also find important information for the configuration of firewalls, Access Control Lists (ACLs), and quality of service (QoS) on a network when an IP Communications solution is implemented.

- Cisco Unified Communications Manager TCP and UDP Port Usage Overview, on page 653
- Port Descriptions, on page 655
- Port References, on page 669

Cisco Unified Communications Manager TCP and UDP Port Usage Overview

Cisco Unified Communications Manager TCP and UDP ports are organized into the following categories:

- Intracluster Ports Between Cisco Unified Communications Manager Servers
- Common Service Ports
- Ports Between Cisco Unified Communications Manager and LDAP Directory
- Web Requests From CCMAdmin or CCMUser to Cisco Unified Communications Manager
- Web Requests From Cisco Unified Communications Manager to Phone
- Signaling, Media, and Other Communication Between Phones and Cisco Unified Communications Manager
- Signaling, Media, and Other Communication Between Gateways and Cisco Unified Communications Manager
- Communication Between Applications and Cisco Unified Communications Manager
- Communication Between CTL Client and Firewalls
- Special Ports on HP Servers

See “Port Descriptions” for port details in each of the above categories.
Cisco has not verified all possible configuration scenarios for these ports. If you are having configuration problems using this list, contact Cisco technical support for assistance.

Port references apply specifically to Cisco Unified Communications Manager. Some ports change from one release to another, and future releases may introduce new ports. Therefore, make sure that you are using the correct version of this document for the version of Cisco Unified Communications Manager that is installed.

While virtually all protocols are bidirectional, directionality from the session originator perspective is presumed. In some cases, the administrator can manually change the default port numbers, though Cisco does not recommend this as a best practice. Be aware that Cisco Unified Communications Manager opens several ports strictly for internal use.

Installing Cisco Unified Communications Manager software automatically installs the following network services for serviceability and activates them by default. Refer to “Intracluster Ports Between Cisco Unified Communications Manager Servers” for details:

- Cisco Log Partition Monitoring (To monitor and purge the common partition. This uses no custom common port.)
- Cisco Trace Collection Service (TCTS port usage)
- Cisco RIS Data Collector (RIS server port usage)
- Cisco AMC Service (AMC port usage)

Configuration of firewalls, ACLs, or QoS will vary depending on topology, placement of telephony devices and services relative to the placement of network security devices, and which applications and telephony extensions are in use. Also, bear in mind that ACLs vary in format with different devices and versions.

You can also configure Multicast Music on Hold (MOH) ports in Cisco Unified Communications Manager. Port values for multicast MOH are not provided because the administrator specifies the actual port values.

The ephemeral port range for the system is 32768 to 61000. For more information, see http://www.cisco.com/c/en/us/support/security/asa-5500-series-next-generation-firewalls/tsd-products-support-series-home.html.

Make sure that you configure your firewall so that connections to port 22 are open, and are not throttled. During the installation of IM and Presence subscriber nodes, multiple connections to the Cisco Unified Communications Manager publisher node are opened in quick succession. Throttling these connections could lead to a failed installation.
# Port Descriptions

## Intracluster Ports Between Cisco Unified Communications Manager Servers

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Endpoint</td>
<td>Unified Communications Manager</td>
<td>514 / UDP</td>
<td>System logging service</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>RTMT</td>
<td>1090, 1099 / TCP</td>
<td>Cisco AMC Service for RTMT performance monitors, data collection, logging, and alerting</td>
</tr>
<tr>
<td>Unified Communications Manager (DB)</td>
<td>Unified Communications Manager (DB)</td>
<td>1500, 1501 / TCP</td>
<td>Database connection (1501 / TCP is the secondary connection)</td>
</tr>
<tr>
<td>Unified Communications Manager (DB)</td>
<td>Unified Communications Manager (DB)</td>
<td>1510 / TCP</td>
<td>CAR IDS DB. CAR IDS engine listens on waiting for connection requests from the clients.</td>
</tr>
<tr>
<td>Unified Communications Manager (DB)</td>
<td>Unified Communications Manager (DB)</td>
<td>1511 / TCP</td>
<td>CAR IDS DB. An alternate port used to bring up a second instance of CAR IDS during upgrade.</td>
</tr>
<tr>
<td>Unified Communications Manager (DB)</td>
<td>Unified Communications Manager (DB)</td>
<td>1515 / TCP</td>
<td>Database replication between nodes during installation</td>
</tr>
<tr>
<td>Cisco Extended Functions (QRT)</td>
<td>Unified Communications Manager (DB)</td>
<td>2552 / TCP</td>
<td>Allows subscribers to receive Cisco Unified Communications Manager database change notification</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>2551 / TCP</td>
<td>Intracluster communication between Cisco Extended Services for Active/Backup determination</td>
</tr>
<tr>
<td>Unified Communications Manager (RIS)</td>
<td>Unified Communications Manager (RIS)</td>
<td>2555 / TCP</td>
<td>Real-time Information Services (RIS) database server</td>
</tr>
<tr>
<td>From (Sender)</td>
<td>To (Listener)</td>
<td>Destination Port</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------------------------</td>
<td>--------------------------------------</td>
<td>------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (RIS)</td>
<td>2556 / TCP</td>
<td>Real-time Information Services (RIS) database client for Cisco RIS</td>
</tr>
<tr>
<td>(RTMT/AMC/SOAP)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (DRS)</td>
<td>4040 / TCP</td>
<td>DRS Primary Agent</td>
</tr>
<tr>
<td>(DRS)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SOAP)</td>
<td>5001 / TCP</td>
<td>This port is used by SOAP monitor for Real Time Monitoring Service.</td>
</tr>
<tr>
<td>(Tomcat)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SOAP)</td>
<td>5002 / TCP</td>
<td>This port is used by SOAP monitor for Performance Monitor Service.</td>
</tr>
<tr>
<td>(Tomcat)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SOAP)</td>
<td>5003 / TCP</td>
<td>This port is used by SOAP monitor for Control Center Service.</td>
</tr>
<tr>
<td>(Tomcat)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SOAP)</td>
<td>5004 / TCP</td>
<td>This port is used by SOAP monitor for Log Collection Service.</td>
</tr>
<tr>
<td>(Tomcat)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Standard CCM Admin Users / Admin</td>
<td>Unified Communications Manager</td>
<td>5005 / TCP</td>
<td>This port is used by SOAP monitor CDROnDemand2 services</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SOAP)</td>
<td>5007 / TCP</td>
<td>SOAP monitor</td>
</tr>
<tr>
<td>(Tomcat)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (TCTS)</td>
<td>Ephemeral / TCP</td>
<td>Cisco Trace Collection Tool Service (TCTS) -- the back end service for</td>
</tr>
<tr>
<td>(RTMT)</td>
<td></td>
<td></td>
<td>RTMT Trace and Log Central (TLC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (TCTS)</td>
<td>7000, 7001, 7002 / TCP</td>
<td>This port is used for communication between Cisco Trace Collection</td>
</tr>
<tr>
<td>(Tomcat)</td>
<td></td>
<td></td>
<td>Tool Service and Cisco Trace Collection servlet.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (CDLM)</td>
<td>8001 / TCP</td>
<td>Client database change notification</td>
</tr>
<tr>
<td>(DB)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SDL)</td>
<td>8002 / TCP</td>
<td>Intracluster communication service</td>
</tr>
<tr>
<td>(SDL)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager (SDL)</td>
<td>8003 / TCP</td>
<td>Intracluster communication service (to CTI)</td>
</tr>
<tr>
<td>(SDL)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>From (Sender)</td>
<td>To (Listener)</td>
<td>Destination Port</td>
<td>Purpose</td>
</tr>
<tr>
<td>-----------------------------------------</td>
<td>----------------------------------------------</td>
<td>------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>CMI Manager</td>
<td>8004 / TCP</td>
<td>Intracluster communication between Cisco Unified Communications Manager and CMI Manager</td>
</tr>
<tr>
<td>Unified Communications Manager (Tomcat)</td>
<td>Unified Communications Manager (Tomcat)</td>
<td>8005 / TCP</td>
<td>Internal listening port used by Tomcat shutdown scripts</td>
</tr>
<tr>
<td>Unified Communications Manager (Tomcat)</td>
<td>Unified Communications Manager (Tomcat)</td>
<td>8080 / TCP</td>
<td>Communication between servers used for diagnostic tests</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>8090</td>
<td>HTTP Port for communication between CuCM and GW (Cayuga interface) for Gateway Recording feature.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager (IPSec)</td>
<td>Unified Communications Manager (IPSec)</td>
<td>8500 / TCP and UDP</td>
<td>Intracluster replication of system data by IPSec Cluster Manager</td>
</tr>
<tr>
<td>Unified Communications Manager (RIS)</td>
<td>Unified Communications Manager (RIS)</td>
<td>8888 - 8889 / TCP</td>
<td>RIS Service Manager status request and reply</td>
</tr>
<tr>
<td>Location Bandwidth Manager (LBM)</td>
<td>Location Bandwidth Manager (LBM)</td>
<td>9004 / TCP</td>
<td>Intracluster communication between LBMs</td>
</tr>
<tr>
<td>Unified Communications Manager Publisher</td>
<td>Unified Communications Manager Subscriber</td>
<td>22 / TCP</td>
<td>Cisco SFTP service. You must open this port when installing a new subscriber.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>8443 / TCP</td>
<td>Allows access to Control Center - Feature and Network service between nodes.</td>
</tr>
</tbody>
</table>
## Common Service Ports

### Table 100: Common Service Ports

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Endpoint</td>
<td>Unified Communications Manager</td>
<td>7</td>
<td>Internet Control Message Protocol (ICMP) This protocol number carries echo-related traffic. It does not constitute a port as indicated in the column heading.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Endpoint</td>
<td></td>
<td>Send the backup data to SFTP server. (DRS Local Agent) Send the CDR data to SFTP server.</td>
</tr>
<tr>
<td>Unified Communications Manager (DRS, CDR)</td>
<td>SFTP server</td>
<td>22/TCP</td>
<td>Send the backup data to SFTP server. (DRS Local Agent) Send the CDR data to SFTP server.</td>
</tr>
</tbody>
</table>
| Endpoint                      | Unified Communications Manager (DHCP Server) | 67/UDP           | Cisco Unified Communications Manager acting as a DHCP server  
  **Note** Cisco does not recommend running DHCP server on Cisco Unified Communications Manager. |
| Unified Communications Manager | DHCP Server               | 68/UDP           | Cisco Unified Communications Manager acting as a DHCP client  
  **Note** Cisco does not recommend running DHCP client on Cisco Unified Communications Manager. Configure Cisco Unified Communications Manager with static IP addresses instead.) |
<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Endpoint or Gateway</td>
<td>Unified Communications Manager</td>
<td>69, 6969, then Ephemeral / UDP</td>
<td>Trivial File Transfer Protocol (TFTP) service to phones and gateways</td>
</tr>
<tr>
<td>Endpoint or Gateway</td>
<td>Unified Communications Manager</td>
<td>6970 / TCP</td>
<td>Trivial File Transfer Protocol (TFTP) between primary and proxy servers. HTTP service from the TFTP server to phones and gateways.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>NTP Server</td>
<td>123 / UDP</td>
<td>Network Time Protocol (NTP)</td>
</tr>
<tr>
<td>SNMP Server</td>
<td>Unified Communications Manager</td>
<td>161 / UDP</td>
<td>SNMP service response (requests from management applications)</td>
</tr>
<tr>
<td>CUCM Server SNMP Primary Agent application</td>
<td>SNMP trap destination</td>
<td>162 / UDP</td>
<td>SNMP traps</td>
</tr>
<tr>
<td>SNMP Server</td>
<td>Unified Communications Manager</td>
<td>199 / TCP</td>
<td>Native SNMP agent listening port for SMUX support</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>DHCP Server</td>
<td>546 / UDP</td>
<td>DHCPv6. DHCP port for IPv6.</td>
</tr>
<tr>
<td>Unified Communications Manager Serviceability</td>
<td>Location Bandwidth Manager (LBM)</td>
<td>5546 / TCP</td>
<td>Enhanced Location CAC Serviceability</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Location Bandwidth Manager (LBM)</td>
<td>5547 / TCP</td>
<td>Call Admission requests and bandwidth deductions</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>6161 / UDP</td>
<td>Used for communication between Primary Agent and Native Agent to process Native agent MIB requests</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>6162 / UDP</td>
<td>Used for communication between Primary Agent and Native Agent to forward notifications generated from Native Agent</td>
</tr>
<tr>
<td>Centralized TFTP</td>
<td>Alternate TFTP</td>
<td>6970 / TCP</td>
<td>Centralized TFTP File Locator Service</td>
</tr>
<tr>
<td>From (Sender)</td>
<td>To (Listener)</td>
<td>Destination Port</td>
<td>Purpose</td>
</tr>
<tr>
<td>--------------------------------------</td>
<td>----------------------------------------</td>
<td>------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>7161 / TCP</td>
<td>Used for communication between SNMP Primary Agent and subagents</td>
</tr>
<tr>
<td>SNMP Server</td>
<td>Unified Communications Manager</td>
<td>7999 / TCP</td>
<td>Cisco Discovery Protocol (CDP) agent communicates with CDP executable</td>
</tr>
<tr>
<td>Endpoint</td>
<td>Unified Communications Manager</td>
<td>443, 8443 / TCP</td>
<td>Used for Cisco User Data Services (UDS) requests</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>9050 / TCP</td>
<td>Service CRS requests through the TAPS residing on Cisco Unified Manager</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>61441 / UDP</td>
<td>Cisco Unified Communications Manager applications send out alarms to this port through UDP. Cisco Unified Communications Manager MIB agent listens on this port and generates SNMP traps per Cisco Unified Communications Manager MIB definition.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>5060, 5061 / TCP</td>
<td>Provide trunk-based SIP services</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>7501</td>
<td>Used by Intercluster Lookup Service (ILS) for certificate based authentication.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Unified Communications Manager</td>
<td>7502</td>
<td>Used by ILS for password based authentication.</td>
</tr>
<tr>
<td>--</td>
<td>--</td>
<td>8000-48200</td>
<td>ASR and ISR G3 platforms default port range.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>16384-32766</td>
<td>ISR G2 platform default port range.</td>
</tr>
</tbody>
</table>
Ports Between Cisco Unified Communications Manager and LDAP Directory

Table 101: Ports Between Cisco Unified Communications Manager and LDAP Directory

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>External Directory</td>
<td>Unified Communications Manager</td>
<td>Ephemeral</td>
<td></td>
</tr>
</tbody>
</table>

Web Requests From CCMAadmin or CCMUser to Cisco Unified Communications Manager

Table 102: Web Requests From CCMAadmin or CCMUser to Cisco Unified Communications Manager

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Browser</td>
<td>Unified Communications Manager</td>
<td>80, 8080 / TCP</td>
<td>Hypertext Transport Protocol (HTTP)</td>
</tr>
<tr>
<td>Browser</td>
<td>Unified Communications Manager</td>
<td>443, 8443 / TCP</td>
<td>Hypertext Transport Protocol over SSL (HTTPS)</td>
</tr>
</tbody>
</table>

Web Requests From Cisco Unified Communications Manager to Phone

Table 103: Web Requests From Cisco Unified Communications Manager to Phone

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unified Communications Manager</td>
<td>Phone</td>
<td>80 / TCP</td>
<td>Hypertext Transport Protocol (HTTP)</td>
</tr>
<tr>
<td>• QRT</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• RTMT</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Find and List Phones page</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• Phone Configuration page</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
## Signaling, Media, and Other Communication Between Phones and Cisco Unified Communications Manager

Table 104: Signaling, Media, and Other Communication Between Phones and Cisco Unified Communications Manager

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>53/ TCP</td>
<td>Session Initiation Protocol (SIP) phones resolve the Fully Qualified Domain Name (FQDN) using a Domain Name System (DNS)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td><strong>Note</strong> By default, some wireless access points block TCP 53 port, which prevents wireless SIP phones from registering when CUCM is configured using FQDN.</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager (TFTP)</td>
<td>69, then Ephemeral / UDP</td>
<td>Trivial File Transfer Protocol (TFTP) used to download firmware and configuration files</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>2000 / TCP</td>
<td>Skinny Client Control Protocol (SCCP)</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>2443 / TCP</td>
<td>Secure Skinny Client Control Protocol (SCCPS)</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>2445 / TCP</td>
<td>Provide trust verification service to endpoints.</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager (CAPF)</td>
<td>3804 / TCP</td>
<td>Certificate Authority Proxy Function (CAPF) listening port for issuing Locally Significant Certificates (LSCs) to IP phones</td>
</tr>
<tr>
<td>From (Sender)</td>
<td>To (Listener)</td>
<td>Destination Port</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>--------------------------------</td>
<td>------------------</td>
<td>-----------------------------------------------------------</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>5060 / TCP and UDP</td>
<td>Session Initiation Protocol (SIP) phone</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Phone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>5061 TCP</td>
<td>Secure Session Initiation Protocol (SIPS) phone</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Phone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager (TFTP)</td>
<td>6970 TCP</td>
<td>HTTP-based download of firmware and configuration files</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager (TFTP)</td>
<td>6971, 6972 / TCP</td>
<td>HTTPS interface to TFTP. Phones use this port to download a secure configuration file from TFTP.</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>8080 / TCP</td>
<td>Phone URLs for XML applications, authentication, directories, services, etc. You can configure these ports on a per-service basis.</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>9443 / TCP</td>
<td>Phone use this port for authenticated contact search.</td>
</tr>
<tr>
<td>Phone</td>
<td>Unified Communications Manager</td>
<td>9444</td>
<td></td>
</tr>
<tr>
<td>IP VMS</td>
<td>Phone</td>
<td>16384 - 32767 / UDP</td>
<td>Real-Time Protocol (RTP), Secure Real-Time Protocol (SRTP)</td>
</tr>
<tr>
<td>Phone</td>
<td>IP VMS</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Note**  
Cisco Unified Communications Manager only uses 24576-32767 although other devices use the full range.
## Signaling, Media, and Other Communication Between Gateways and Cisco Unified Communications Manager

### Table 105: Signaling, Media, and Other Communication Between Gateways and Cisco Unified Communications Manager

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>47, 50, 51</td>
<td>Generic Routing Encapsulation (GRE), Encapsulating Security Payload (ESP), Authentication Header (AH). These protocols numbers carry encrypted IPSec traffic. They do not constitute a port as indicated in the column heading.</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td>Internet Key Exchange (IKE) for IP Security protocol (IPSec) establishment</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>500 / UDP</td>
<td>Trivial File Transfer Protocol (TFTP)</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td>Port mapping service. Only used in the CIME off-path deployment model.</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager (TFTP)</td>
<td>69, then Ephemeral / UDP</td>
<td>Gatekeeper (H.225) RAS</td>
</tr>
<tr>
<td>Unified Communications Manager with Cisco Intercompany Media Engine (CIME) trunk</td>
<td>CIME ASA</td>
<td>1024-65535 / TCP</td>
<td>H.225 signaling services for H.323 gateways and Intercluster Trunk (ICT)</td>
</tr>
<tr>
<td>Gatekeeper</td>
<td>Unified Communications Manager</td>
<td>1719 / UDP</td>
<td>H.225 signaling services on gatekeeper-controlled trunk</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>1720 / TCP</td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>Ephemeral / TCP</td>
<td></td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
<tr>
<td>From (Sender)</td>
<td>To (Listener)</td>
<td>Destination Port</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>--------------------------------</td>
<td>------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>Ephemeral / TCP</td>
<td>H.245 signaling services for establishing voice, video, and data</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td><strong>Note</strong> The H.245 port used by the remote system depends on the type of gateway. For IOS gateways, the H.245 port range is from 11000 to 65535.</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>2000 / TCP</td>
<td>Skinny Client Control Protocol (SCCP)</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>2001 / TCP</td>
<td>Upgrade port for 6608 gateways with Cisco Unified Communications Manager deployments</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>2002 / TCP</td>
<td>Upgrade port for 6624 gateways with Cisco Unified Communications Manager deployments</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>2427 / UDP</td>
<td>Media Gateway Control Protocol (MGCP) gateway control</td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>2428 / TCP</td>
<td>Media Gateway Control Protocol (MGCP) backhaul</td>
</tr>
<tr>
<td>--</td>
<td>--</td>
<td>4000 - 4005 / TCP</td>
<td>These ports are used as phantom Real-Time Transport Protocol (RTP) and Real-Time Transport Control Protocol (RTCP) ports for audio, video and data channel when Cisco Unified Communications Manager does not have ports for these media.</td>
</tr>
</tbody>
</table>
## Communication Between Applications and Cisco Unified Communications Manager

### Table 106: Communication Between Applications and Cisco Unified Communications Manager

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>5060 / TCP and UDP</td>
<td>Session Initiation Protocol (SIP) gateway and Intercluster Trunk (ICT)</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>5061 / TCP</td>
<td>Secure Session Initiation Protocol (SIPS) gateway and Intercluster Trunk (ICT)</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Gateway</td>
<td>Unified Communications Manager</td>
<td>16384 - 32767 / UDP</td>
<td>Real-Time Protocol (RTP), Secure Real-Time Protocol (SRTP)</td>
</tr>
<tr>
<td>Unified Communications Manager</td>
<td>Gateway</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Note**: Cisco Unified Communications Manager only uses 24576-32767 although other devices use the full range.
<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unified Communications Manager Assistant Console</td>
<td>Unified Communications Manager</td>
<td>2912 / TCP</td>
<td>Cisco Unified Communications Manager Assistant server (formerly IPMA)</td>
</tr>
<tr>
<td>Unified Communications Manager Attendant Console</td>
<td>Unified Communications Manager</td>
<td>1103 -1129 / TCP</td>
<td>Cisco Unified Communications Manager Attendant Console (AC) JAVA RMI Registry server</td>
</tr>
<tr>
<td>Unified Communications Manager Attendant Console</td>
<td>Unified Communications Manager</td>
<td>1101 / TCP</td>
<td>RMI server sends RMI callback messages to clients on these ports.</td>
</tr>
<tr>
<td>Unified Communications Manager Attendant Console</td>
<td>Unified Communications Manager</td>
<td>1102 / TCP</td>
<td>Attendant Console (AC) RMI server bind port -- RMI server sends RMI messages on these ports.</td>
</tr>
<tr>
<td>Unified Communications Manager Attendant Console</td>
<td>Unified Communications Manager</td>
<td>3223 / UDP</td>
<td>Cisco Unified Communications Manager Attendant Console (AC) server line state port receives ping and registration message from, and sends line states to, the attendant console server.</td>
</tr>
<tr>
<td>Unified Communications Manager Attendant Console</td>
<td>Unified Communications Manager</td>
<td>3224 / UDP</td>
<td>Cisco Unified Communications Manager Attendant Console (AC) clients register with the AC server for line and device state information.</td>
</tr>
<tr>
<td>Unified Communications Manager Attendant Console</td>
<td>Unified Communications Manager</td>
<td>4321 / UDP</td>
<td>Cisco Unified Communications Manager Attendant Console (AC) clients register to the AC server for call control.</td>
</tr>
<tr>
<td>Unified Communications Manager with SAF/CCD</td>
<td>IOS Router running SAF image</td>
<td>5050 / TCP</td>
<td>Multi-Service IOS Router running EIGRP/SAF Protocol.</td>
</tr>
</tbody>
</table>
### Communication Between CTL Client and Firewalls

**Table 107: Communication Between CTL Client and Firewalls**

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>CTL Client</td>
<td>TLS Proxy Server</td>
<td>2444 / TCP</td>
<td>Certificate Trust List (CTL) provider listening service in an ASA firewall</td>
</tr>
</tbody>
</table>

### Special Ports on HP Servers

**Table 108: Special Ports on HP Servers**

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Destination Port</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Endpoint</td>
<td>HP SIM</td>
<td>2301 / TCP</td>
<td>HTTP port to HP agent</td>
</tr>
<tr>
<td>Endpoint</td>
<td>HP SIM</td>
<td>2381 / TCP</td>
<td>HTTPS port to HP agent</td>
</tr>
<tr>
<td>Endpoint</td>
<td>Compaq Management Agent</td>
<td>25375, 25376, 25393 / UDP</td>
<td>COMPAQ Management Agent extension (cmaX)</td>
</tr>
<tr>
<td>Endpoint</td>
<td>HP SIM</td>
<td>50000 - 50004 / TCP</td>
<td>HTTPS port to HP SIM</td>
</tr>
</tbody>
</table>
Port References

Firewall Application Inspection Guides
ASA Series reference information
PIX Application Inspection Configuration Guides
FWSM 3.1 Application Inspection Configuration Guide

IETF TCP/UDP Port Assignment List
Internet Assigned Numbers Authority (IANA) IETF assigned Port List
http://www.iana.org/assignments/port-numbers

IP Telephony Configuration and Port Utilization Guides
Cisco CRS 4.0 (IP IVR and IPCC Express) Port Utilization Guide
Port Utilization Guide for Cisco ICM/IPCC Enterprise and Hosted Editions
Cisco Unified Communications Manager Express Security Guide to Best Practices
Cisco Unity Express Security Guide to Best Practices

VMware Port Assignment List
TCP and UDP Ports for vCenter Server, ESX hosts, and Other Network Components Management Access
Port Usage Information for the IM and Presence Service

IM and Presence Service Port Usage Overview

This document provides a list of the TCP and UDP ports that the IM and Presence Service uses for intracluster connections and for communications with external applications or devices. It provides important information for the configuration of firewalls, Access Control Lists (ACLs), and quality of service (QoS) on a network when an IP Communications solution is implemented.

Cisco has not verified all possible configuration scenarios for these ports. If you are having configuration problems using this list, contact Cisco technical support for assistance.

While virtually all protocols are bidirectional, this document gives directionality from the session originator perspective. In some cases, the administrator can manually change the default port numbers, though Cisco does not recommend this as a best practice. Be aware that the IM and Presence Service opens several ports strictly for internal use.

Ports in this document apply specifically to the IM and Presence Service. Some ports change from one release to another, and future releases may introduce new ports. Therefore, make sure that you are using the correct version of this document for the version of IM and Presence Service that is installed.

Configuration of firewalls, ACLs, or QoS will vary depending on topology, placement of devices and services relative to the placement of network security devices, and which applications and telephony extensions are in use. Also, bear in mind that ACLs vary in format with different devices and versions.

Information Collated in Table

This table defines the information collated in each of the tables in this document.
Table 109: Definition of Table Information

<table>
<thead>
<tr>
<th>Table Heading</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>From</td>
<td>The client sending requests to this port</td>
</tr>
<tr>
<td>To</td>
<td>The client receiving requests on this port</td>
</tr>
<tr>
<td>Role</td>
<td>A client or server application or process</td>
</tr>
<tr>
<td>Protocol</td>
<td>Either a Session-layer protocol used for establishing and ending communications, or an Application-layer protocol used for request and response transactions</td>
</tr>
<tr>
<td>Transport Protocol</td>
<td>A Transport-layer protocol that is connection-oriented (TCP) or connectionless (UDP)</td>
</tr>
<tr>
<td>Destination / Listener</td>
<td>The port used for receiving requests</td>
</tr>
<tr>
<td>Source / Sender</td>
<td>The port used for sending requests</td>
</tr>
</tbody>
</table>

### IM and Presence Service Port List

The following tables show the ports that the IM and Presence Service uses for intracluster and intercluster traffic.

Table 110: IM and Presence Service Ports - SIP Proxy Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Gateway</td>
<td>IM and Presence</td>
<td>SIP</td>
<td>TCP/UDP</td>
<td>5060</td>
<td>Ephemeral</td>
<td>Default SIP Proxy UDP and TCP Listener</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>SIP Gateway</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>SIP</td>
<td>TLS</td>
<td>5061</td>
<td>Ephemeral</td>
<td>TLS Server Authentication listener port</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>SIP</td>
<td>TLS</td>
<td>5062</td>
<td>Ephemeral</td>
<td>TLS Mutual Authentication listener port</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>SIP</td>
<td>UDP / TCP</td>
<td>5049</td>
<td>Ephemeral</td>
<td>Internal port. Localhost traffic only.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>HTTP</td>
<td>TCP</td>
<td>8081</td>
<td>Ephemeral</td>
<td>Used for HTTP requests from the Config Agent to indicate a change in configuration.</td>
</tr>
</tbody>
</table>
**Table 111: IM and Presence Service Ports - Presence Engine Requests**

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Third-party Client</td>
<td>IM and Presence</td>
<td>HTTP</td>
<td>TCP</td>
<td>8082</td>
<td>Ephemeral</td>
<td>Default IM and Presence HTTP Listener. Used for Third-Party Clients to connect</td>
</tr>
<tr>
<td>Third-party Client</td>
<td>IM and Presence</td>
<td>HTTPS</td>
<td>TLS / TCP</td>
<td>8083</td>
<td>Ephemeral</td>
<td>Default IM and Presence HTTPS Listener. Used for Third-Party Clients to connect</td>
</tr>
</tbody>
</table>

**Table 112: IM and Presence Service Ports - Cisco Tomcat WebRequests**

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Browser</td>
<td>IM and Presence</td>
<td>HTTPS</td>
<td>TCP</td>
<td>8080</td>
<td>Ephemeral</td>
<td>Used for web access</td>
</tr>
<tr>
<td>Browser</td>
<td>IM and Presence</td>
<td>AXL / HTTPS</td>
<td>TLS / TCP</td>
<td>8443</td>
<td>Ephemeral</td>
<td>Provides database and serviceability access via SOAP</td>
</tr>
<tr>
<td>Browser</td>
<td>IM and Presence</td>
<td>HTTPS</td>
<td>TLS / TCP</td>
<td>8443</td>
<td>Ephemeral</td>
<td>Provides access to Web administration</td>
</tr>
<tr>
<td>Browser</td>
<td>IM and Presence</td>
<td>HTTPS</td>
<td>TLS / TCP</td>
<td>8443</td>
<td>Ephemeral</td>
<td>Provides access to User option pages</td>
</tr>
</tbody>
</table>
### Table 113: IM and Presence Service Ports - External Corporate Directory Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Browser</td>
<td>IM and Presence</td>
<td>SOAP</td>
<td>TLS / TCP</td>
<td>8443</td>
<td>Ephemeral</td>
<td>Provides access to Cisco Unified Personal Communicator, Cisco Unified Mobility Advantage, and third-party API clients via SOAP</td>
</tr>
</tbody>
</table>

**Table 114: IM and Presence Service Ports - Configuration Requests**

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence (Config Agent)</td>
<td>IM and Presence (Config Agent)</td>
<td>TCP</td>
<td>TCP</td>
<td>8600</td>
<td>Ephemeral</td>
<td>Config Agent heartbeat port</td>
</tr>
</tbody>
</table>

**Ephemeral** allows the Directory protocol to integrate with the external Corporate Directory. The LDAP port depends on the Corporate Directory (389 is the default). In case of Netscape Directory, customer can configure different port to accept LDAP traffic.

**Ephemeral** allows LDAP to communicate between IM&P and the LDAP server for authentication.
### Table 115: IM and Presence Service Ports - Certificate Manager Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>Certificate Manager</td>
<td>TCP</td>
<td>TCP</td>
<td>7070</td>
<td>Ephemeral</td>
<td>Internal port - Localhost traffic only</td>
</tr>
</tbody>
</table>

### Table 116: IM and Presence Service Ports - IDS Database Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence(Database)</td>
<td>IM and Presence (Database)</td>
<td>TCP</td>
<td>TCP</td>
<td>1500</td>
<td>Ephemeral</td>
<td>Internal port for Database clients. Localhost traffic only.</td>
</tr>
<tr>
<td>IM and Presence(Database)</td>
<td>IM and Presence (Database)</td>
<td>TCP</td>
<td>TCP</td>
<td>1501</td>
<td>Ephemeral</td>
<td>Internal port - this is an alternate port to bring up a second instance of IDS during upgrade. Localhost traffic only.</td>
</tr>
<tr>
<td>IM and Presence(Database)</td>
<td>IM and Presence (Database)</td>
<td>XML</td>
<td>TCP</td>
<td>1515</td>
<td>Ephemeral</td>
<td>Internal port. Localhost traffic only. DB replication port</td>
</tr>
</tbody>
</table>

### Table 117: IM and Presence Service Ports - IPSec Manager Request

<table>
<thead>
<tr>
<th>From Sender</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence (IPSec)</td>
<td>IM and Presence (IPSec)</td>
<td>Proprietary</td>
<td>UDP/TCP</td>
<td>8500</td>
<td>8500</td>
<td>Internal port - cluster manager port used by the ipsec_mngr daemon for cluster replication of platform data (hosts) certs</td>
</tr>
</tbody>
</table>

### Table 118: IM and Presence Service Ports - DRF Master Agent Server Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence (DRF)</td>
<td>IM and Presence (DRF)</td>
<td>TCP</td>
<td>TCP</td>
<td>4040</td>
<td>Ephemeral</td>
<td>DRF Master Agent server port, which accepts connections from Local Agent, GUI, and CLI</td>
</tr>
</tbody>
</table>
### Table 119: IM and Presence Service Ports - RISDC Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence (RIS)</td>
<td>IM and Presence (RIS)</td>
<td>TCP</td>
<td>TCP</td>
<td>2555</td>
<td>Ephemeral</td>
<td>Real-time Information Services (RIS) database server. Connects to other RISDC services in the cluster to provide clusterwide real-time information.</td>
</tr>
<tr>
<td>IM and Presence (RTMT/AMC/SOAP)</td>
<td>IM and Presence (RIS)</td>
<td>TCP</td>
<td>TCP</td>
<td>2556</td>
<td>Ephemeral</td>
<td>Real-time Information Services (RIS) database client for Cisco RIS. Allows RIS client connection to retrieve real-time information.</td>
</tr>
<tr>
<td>IM and Presence (RIS)</td>
<td>IM and Presence (RIS)</td>
<td>TCP</td>
<td>TCP</td>
<td>8889</td>
<td>8888</td>
<td>Internal port. Localhost traffic only. Used by RISDC (System Access) to link to servM via TCP for service status request and reply.</td>
</tr>
</tbody>
</table>

### Table 120: IM and Presence Service Ports - SNMP Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>SNMP Server</td>
<td>IM and Presence</td>
<td>SNMP</td>
<td>UDP</td>
<td>161, 8161</td>
<td>Ephemeral</td>
<td>Provides services for SNMP-based management applications.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>SNMP</td>
<td>UDP</td>
<td>6162</td>
<td>Ephemeral</td>
<td>Native SNMP agent that listens for requests forwarded by SNMP master agents.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>SNMP</td>
<td>UDP</td>
<td>6161</td>
<td>Ephemeral</td>
<td>SNMP Master agent that listens for traps from the native SNMP agent, and forwards to management applications.</td>
</tr>
<tr>
<td>SNMP Server</td>
<td>IM and Presence</td>
<td>TCP</td>
<td>TCP</td>
<td>7999</td>
<td>Ephemeral</td>
<td>Used as a socket for the cdp agent to communicate with the cdp binary.</td>
</tr>
</tbody>
</table>
### Table 121: IM and Presence Service Ports - Racoon Server Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>TCP</td>
<td>TCP</td>
<td>7161</td>
<td>Ephemeral</td>
<td>Used for communication between the SNMP Master agent and subagents</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>SNMP Trap Monitor</td>
<td>SNMP</td>
<td>UDP</td>
<td>162</td>
<td>Ephemeral</td>
<td>Sends SNMP traps to management applications</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>SNMP</td>
<td>UDP</td>
<td>Configurable</td>
<td>61441</td>
<td>Internal SNMP trap receiver</td>
</tr>
</tbody>
</table>

### Table 122: IM and Presence Service Ports - System Service Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway</td>
<td>IM and Presence</td>
<td>Ipcsec</td>
<td>UDP</td>
<td>500</td>
<td>Ephemeral</td>
<td>Enables Internet Security Association and the Key Management Protocol</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>Gateway</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Table 123: IM and Presence Service Ports - DNS Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>DNS Server</td>
<td>DNS</td>
<td>UDP</td>
<td>53</td>
<td>Ephemeral</td>
<td>The port that DNS server listen on for IM and Presence DNS queries.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>To: DNS Server</td>
</tr>
</tbody>
</table>
### Table 124: IM and Presence Service Ports - SSH/SFTP Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>Endpoint</td>
<td>SSH / SFTP</td>
<td>TCP</td>
<td>22</td>
<td>Ephemeral</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Used by many applications to get command line access to the server. Also used between nodes for certificate and other file exchanges (sftp)</td>
</tr>
</tbody>
</table>

### Table 125: IM and Presence Service Ports - ICMP Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>Cisco Unified Communications Manager</td>
<td>ICMP</td>
<td>IP</td>
<td>Not Applicable</td>
<td>Ephemeral</td>
</tr>
<tr>
<td></td>
<td>IM and Presence</td>
<td></td>
<td></td>
<td></td>
<td>Internet Control Message Protocol (ICMP). Used to communicate with the Cisco Unified Communications Manager server</td>
</tr>
</tbody>
</table>

### Table 126: IM and Presence Service Ports - NTP Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>NTP Server</td>
<td>NTP</td>
<td>UDP</td>
<td>123</td>
<td>Ephemeral</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Cisco Unified Communications Manager is the acting NTP server. Used by subscriber nodes to synchronize time with the publisher node.</td>
</tr>
</tbody>
</table>
### Table 127: IM and Presence Service Ports - Microsoft Exchange Notify Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Microsoft Exchange</td>
<td>IM and Presence</td>
<td>HTTP (HTTP)</td>
<td>1) WebDAV - HTTP /UDP/IP notifications 2) EWS - HTTP/TCP /IP SOAP notifications</td>
<td>IM and Presence server port (default 50020)</td>
<td>Ephemeral</td>
<td>Microsoft Exchange uses this port to send notifications (using NOTIFY message) to indicate a change to a particular subscription identifier for calendar events. Used to integrate with any Exchange server in the network configuration. Both ports are created. The kind of messages that are sent depend on the type of Calendar Presence Backend gateway(s) that are configured.</td>
</tr>
</tbody>
</table>

### Table 128: IM and Presence Service Ports - SOAP Services Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence (Tomcat)</td>
<td>IM and Presence (SOAP)</td>
<td>TCP</td>
<td>TCP</td>
<td>5007</td>
<td>Ephemeral</td>
<td>SOAP monitor port</td>
</tr>
</tbody>
</table>

### Table 129: IM and Presence Service Ports - AMC RMI Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
</table>
### Table 130: IM and Presence Service Ports - XCP Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>XMPP Client</td>
<td>IM and Presence</td>
<td>TCP</td>
<td>TCP</td>
<td>5222</td>
<td>Ephemeral</td>
<td>Client access port</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>TCP</td>
<td>TCP</td>
<td>5269</td>
<td>Ephemeral</td>
<td>Server to Server connection (S2S) port</td>
</tr>
<tr>
<td>Third-party BOSH client</td>
<td>IM and Presence</td>
<td>TCP</td>
<td>TCP</td>
<td>7335</td>
<td>Ephemeral</td>
<td>HTTP listening port used by the XCP Web Connection Manager for BOSH third-party API connections</td>
</tr>
<tr>
<td>IM and Presence (XCP Services)</td>
<td>IM and Presence (XCP Router)</td>
<td>TCP</td>
<td>TCP</td>
<td>7400</td>
<td>Ephemeral</td>
<td>XCP Router Master Accept Port. XCP services that connect to the router from an Open Port Configuration (for example XCP Authentication Component Service) typically connect on this port.</td>
</tr>
<tr>
<td>IM and Presence (XCP Router)</td>
<td>IM and Presence (XCP Router)</td>
<td>UDP</td>
<td>UDP</td>
<td>5353</td>
<td>Ephemeral</td>
<td>MDNS port. XCP routers in a cluster use this port to discover each other.</td>
</tr>
<tr>
<td>IM and Presence (XCP Router)</td>
<td>IM and Presence (XCP Router)</td>
<td>TCP</td>
<td>TCP</td>
<td>7336</td>
<td>HTTPS</td>
<td>MFT File transfer (On-Premises only).</td>
</tr>
</tbody>
</table>

### Table 131: IM and Presence Service Ports - External Database (PostgreSQL) Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>PostgreSQL database</td>
<td>TCP</td>
<td>TCP</td>
<td>5432&lt;sup&gt;2&lt;/sup&gt;</td>
<td>Ephemeral</td>
<td>PostgreSQL database listening port</td>
</tr>
</tbody>
</table>

<sup>2</sup> This is the default port, however you can configure the PostgreSQL database to listen on any port.
<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence (Server Recovery Manager)</td>
<td>IM and Presence (Server Recovery Manager)</td>
<td>TCP</td>
<td>TCP</td>
<td>20075</td>
<td>Ephemeral</td>
<td>The port that Cisco Server Recovery Manager uses to provide admin rpc requests.</td>
</tr>
<tr>
<td>IM and Presence (Server Recovery Manager)</td>
<td>IM and Presence (Server Recovery Manager)</td>
<td>UDP</td>
<td>UDP</td>
<td>22001</td>
<td>Ephemeral</td>
<td>The port that Cisco Server Recovery Manager uses to communicate with its peer.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6603*</td>
<td>Ephemeral</td>
<td>Cisco Presence Datastore</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6604*</td>
<td>Ephemeral</td>
<td>Cisco Login Datastore</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6605*</td>
<td>Ephemeral</td>
<td>Cisco SIP Registration Datastore</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>9003</td>
<td>Ephemeral</td>
<td>Cisco Presence Datastore dual node presence redundancy group replication.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>9004</td>
<td>Ephemeral</td>
<td>Cisco Login Datastore dual node presence redundancy group replication.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>9005</td>
<td>Ephemeral</td>
<td>Cisco SIP Registration Datastore dual node presence redundancy group replication.</td>
</tr>
</tbody>
</table>

* If you want to run the Administration CLI Diagnostic Utility, using the `utils imdb_replication status` command, these ports must be open on all firewalls that are configured between IM and Presence Service nodes in the cluster. This setup is not required for normal operation.
### Table 134: IM and Presence Service Ports - In Memory Database SQL Messages

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6603</td>
<td>Ephemeral</td>
<td>Cisco Presence Datastore SQL Queries.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6604</td>
<td>Ephemeral</td>
<td>Cisco Login Datastore SQL Queries.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6605</td>
<td>Ephemeral</td>
<td>Cisco SIP Registration Datastore SQL Queries.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6606</td>
<td>Ephemeral</td>
<td>Cisco Route Datastore SQL Queries.</td>
</tr>
</tbody>
</table>

### Table 135: IM and Presence Service Ports - In Memory Database Notification Messages

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6609</td>
<td>Ephemeral</td>
<td>Cisco SIP Registration Datastore XML-based change notification.</td>
</tr>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>Proprietary</td>
<td>TCP</td>
<td>6610</td>
<td>Ephemeral</td>
<td>Cisco Route Datastore XML-based change notification.</td>
</tr>
</tbody>
</table>

### Table 136: IM and Presence Service Ports - Force Manual Sync/X.509 Certificate Update Requests

<table>
<thead>
<tr>
<th>From (Sender)</th>
<th>To (Listener)</th>
<th>Protocol</th>
<th>Transport Protocol</th>
<th>Destination / Listener</th>
<th>Source / Sender</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>IM and Presence</td>
<td>IM and Presence</td>
<td>TCP</td>
<td>TCP</td>
<td>37239</td>
<td>Ephemeral</td>
<td>Cisco Intercluster Sync Agent service uses this port to establish a socket connection for handling commands.</td>
</tr>
</tbody>
</table>

See the *Cisco Unified Serviceability Administration Guide* for information about SNMP.