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PART I

Getting Started

• Feature Configuration Overview, on page 1
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CHAPTER 1

Feature Configuration Overview

- About the Feature Configuration Guide, on page 1
- Generate a Phone Feature List, on page 1

About the Feature Configuration Guide

This guide provides information about the tasks that you need to complete in order to configure features on the Unified Communications Manager system. Use this guide after you have configured the call control system, which includes "day 1" configurations such as inbound and outbound calling, dial plans, and network resources. For information about configuring the call control system, see System Configuration Guide for Cisco Unified Communications Manager.

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.
Generate a Phone Feature List
Configuration Tools

- About the Feature Configuration Guide, on page 3
- Configuration Tools Overview, on page 3
- Generate a Phone Feature List, on page 5

About the Feature Configuration Guide

This guide provides information about the tasks that you need to complete in order to configure features on the Unified Communications Manager system. Use this guide after you have configured the call control system, which includes "day 1" configurations such as inbound and outbound calling, dial plans, and network resources. For information about configuring the call control system, see System Configuration Guide for Cisco Unified Communications Manager.

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.

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

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Start your preferred operating system browser.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the address bar of the web browser, enter the following case-sensitive URL:</td>
</tr>
<tr>
<td></td>
<td>https://&lt;Unified CM-server-name&gt;:{8443}/ccmadmin/showHome.do</td>
</tr>
<tr>
<td></td>
<td>where: &lt;Unified CM-server-name&gt; equals the name or IP address of the server</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>You can optionally specify a port number.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 3</th>
<th>A Security Alert dialog box displays. Click the appropriate button.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 4</td>
<td>At the main Cisco Unified CM Administration window, enter the username and password that you specified during Unified Communications Manager installation and click <strong>Login</strong>. (If you want to clear the content of both fields, click <strong>Reset</strong>.)</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>For security purposes, Cisco Unified Communications Manager Administration logs you out after 30 minutes of inactivity, and you must log back in.</td>
</tr>
</tbody>
</table>

**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**

**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.

**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.

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.
PART II

Remote Worker Features

• Cisco Unified Mobility, on page 9
• Device Mobility, on page 43
• Extend and Connect, on page 55
• Remote Worker Emergency Calling, on page 65
• Configure Mobile and Remote Access, on page 69
Cisco Unified Mobility Overview

Cisco Unified Mobility offers a set of mobility-related features that allow users to interact with Unified Communications applications no matter where they may be, or which device they are using. Whether the device you are using is a home office phone, a dual-mode Cisco Jabber on iPhone or Android client over a WiFi connection, or a mobile phone from another cellular provider, you can still access Unified Communications features and have the call be anchored in the enterprise.

For example, you can answer a call that is directed to your enterprise number from any of your configured phones and then transfer the call to your mobile phone, allowing you to continue an in-progress conversation as you are leaving the office.

Benefits of Cisco Unified Mobility

Most of the mobility features offer call anchoring within the enterprise—even if the call is placed to or from a mobile device, the call is routed through an enterprise gateway.

This provides the following benefits:

- Single enterprise phone number and voicemail for all business calls, regardless of which device you are using, and whether you are in the office or out of the office.

- Ability to extend business calls to a mobile device and have the call still be handled as if it were your office phone.

- Calls placed from mobile devices are anchored to the enterprise and routed through an enterprise gateway. This provides access to UC mid-call features, centralized billing and call detail records, and potential cost savings from avoiding expensive cellular networks.

- Ability to roam from one network to another and have the call not be dropped.
# Mobility Features

Cisco Unified Mobility offers the following mobility-related features:

<table>
<thead>
<tr>
<th>Mobility Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Single Number Reach</td>
<td>Provides you with a single enterprise phone number and voicemail by which you can be reached, regardless of whether you are in the office or outside the office. When someone dials your enterprise number, you can answer the call from your deskphone, or from any of your configured remote destinations (for example, a home office phone, a dual-mode Cisco Jabber on iPhone or Android client, and even a mobile phone from another provider).</td>
</tr>
<tr>
<td>Move to Mobile</td>
<td>Allows you to transfer an active call from your desk-phone to a mobile device that is configured as a remote destination by pressing the Mobility soft-key on your Cisco IP Phone. It is associated with Single Number Reach as a part of the Remote Destination configuration.</td>
</tr>
<tr>
<td></td>
<td>Similar to the Move to Mobile option is the Desk Pickup option, which fits the example where you are on a mobile call and are just arriving at the office. You can hang up on the call on your mobile device and immediately resume the call by picking up your deskphone before the <strong>Maximum Wait Time for Desk Pickup</strong> timer expires (the default is 10 seconds). This option is enabled as part of your Single Number Reach configuration.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> You can also use the Enterprise Feature Access code and the Session Handoff codes to transfer calls between your remote destinations and desk phone.</td>
</tr>
<tr>
<td>Mobile Voice Access</td>
<td>Allows you to place calls from any remote phone and have the call be anchored in the enterprise and presented to the called party as if you had called from your office phone. When using this feature, you must dial in to a system IVR from your mobile device. After authenticating you, and prompting you for the call destination, the system places the call as if you had called from your enterprise phone. You can also use <strong>Mobile Voice Access</strong> prompts to enable or disable <strong>Single Number Reach</strong> for a remote destination.</td>
</tr>
<tr>
<td>Enterprise Feature Access</td>
<td>Provides two-stage dialing from a configured remote destination and have the call that is presented to the called party as if it originated from your desk phone. Unlike <strong>Mobile Voice Access</strong>, to use <strong>Enterprise Feature Access</strong>, you must be dialing from one of your configured remote destinations. <strong>Enterprise Feature Access</strong> also allows you to access mid-call features while on a call from a remote destination. You can access mid-call features by sending DTMF digits that represent the codes for the various features such as Hold, Exclusive Hold, Transfer.</td>
</tr>
</tbody>
</table>
### Mobility Feature

<table>
<thead>
<tr>
<th>Mobility Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intelligent Session Control</td>
<td>Enables automatic call anchoring for enterprise-originated calls that are placed directly to configured remote destination numbers (for example, an enterprise-originated call to a cell phone number that is configured as a remote destination). By configuring a service parameter, you can have the system redirect those calls automatically to the associated enterprise number, providing cost savings and added UC functionality.</td>
</tr>
<tr>
<td>Dual-Mode Phones</td>
<td>Cisco Jabber on iPhone and Android clients can be provisioned as dual-mode devices. <strong>Dual-Mode phones</strong> have the capability of connecting over Wifi or through cellular networks. When the client is within the enterprise network, Cisco Jabber can register to Unified Communications Manager over Wifi, and has UC calling and instant messaging functionality. If you configure a mobile identity with the phone number of the mobile device, allowing the call to be transferred from Jabber to the cellular device when leaving the enterprise network. <strong>Note</strong> An added feature that is available to Cisco Jabber mobile clients is Mobile and Remote Access, which allows Cisco Jabber clients to connect to data networks when outside of the enterprise network. For more information, see &quot;Configure Mobile and Remote Access&quot; section in Feature Configuration Guide for Cisco Unified Communications Manager.</td>
</tr>
</tbody>
</table>

### Cisco Unified Mobility Prerequisites

Refer to the following prerequisites:

- Enabling Mobility features requires proper planning to ensure that your dial plan and call routing configuration can handle the deployment needs. For more information, see "Mobile Collaboration" section in the Cisco Collaboration System Solution Reference Network Designs guide.

- For information on which Cisco IP Phones support Mobility feature, see Generate a Phone Feature List, on page 1.
  - For a list of Cisco IP Phones that support the Mobility softkey, run a report for the **Mobility** feature.
  - For a list of supported dual-mode phones, run a report for the **Dual-Mode** feature.

- If you are deploying Mobile Voice Access and you want to make additional locales available to your system (if you want to use non-English phone locales or country-specific tones), you can download the locale installers from cisco.com and install them through the Cisco Unified OS Administration interface. For more information on installing locales, see Installation Guide for Cisco Unified Communications Manager and the IM and Presence Service.

- Configure Self-Provisioning so that phone users can provision their own Cisco Jabber clients and remote destinations. For more information, see "Configure Self Provisioning" and "Provisioning End Users" section in the System Configuration Guide for Cisco Unified Communications Manager.
The Cisco mobility solution is verified with only Cisco equipment. This solution may also work with other third-party PSTN gateways and Session Border Controllers (SBCs), but the features might not work as described here. If you are using this solution with third-party PSTN gateways or SBCs, Cisco technical support may not be able to resolve problems that you encounter.

## Cisco Unified Mobility Configuration Task Flow

Complete these tasks to configure Mobility features for your deployment.

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 1 | Perform one of the following:  
  • Configure a Mobility User, on page 13  
  • Configure Mobility Users through Bulk Administration, on page 14  
  • Provision Mobility Users Through LDAP, on page 14 | Adds mobility features for an individual end user.  
  Configures Mobility features for a large number of existing end users, use the Bulk Administration Tool.  
  Provisions new users with mobility functionality, you can use a feature group template and LDAP sync. |
| Step 2 | Configure Mobility for IP Phones, on page 15 | Configures Cisco IP Phones for Mobility including setting up the Single Number Reach (SNR) and Move to Mobile features. This allows enterprise phone users to extend enterprise calls to a wide range of mobile devices, including a home office phone or a mobile phone. |
| Step 3 | Configure Mobile Voice Access, on page 20 | **Optional.** Provides a system IVR so that mobile users can call from any mobile device and have the call that is presented to the called party as if the caller were dialing from their enterprise desk phone. |
| Step 4 | Configure Enterprise Feature Access, on page 27 | **Optional.** Provides two-stage dialing from a configured remote destination and have the call that is presented to the called party as if it originated from a desk phone. This feature also allows you to access mid-call features while on a call from a remote destination. |
| Step 5 | Configure Intelligent Session Control, on page 28 | Configure the system so that inbound calls to a remote destination are rerouted to an associated enterprise, if one is available. This provides automatic call anchoring within the enterprise |
### Command or Action | Purpose
---|---

**Step 6**  Configure Mobility Service Parameters, on page 29  
*Optional.* Configure optional mobility-related service parameters if you want to change the behavior of Cisco Unified Mobility.

**Step 7**  Configure Cisco Jabber Dual-Mode, on page 29  
Configure Cisco Jabber for mobility so your users can access enterprise communications features through a Jabber client on their smartphone.

**Step 8**  Configure Other Dual-Mode Devices, on page 30  
Complete this task flow if you want to deploy other dual-mode devices, such as FMC or IMS clients that can connect through Wi-Fi.

### Configure a Mobility User

Configure an end user with the mobility feature.

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose **User Management > End User**.

**Step 2**  In **Find and List Users** window, perform one of the following tasks:
- Click **Find** and select an existing user to modify the settings.
- Click **Add New** to configure a new user.

**Step 3**  Configure values for the following fields:
- **User ID**
- **Last Name**

**Step 4**  In the **Mobility Information** section, complete the following fields:

a)  Check the **Enable Mobility** check box.

b)  **Optional.** Check the **Enable Mobile Voice Access** check box to allow this user to use Mobile Voice Access.

c)  In the **Maximum Wait Time for Desk Pickup** field, enter a value in milliseconds. After hanging up a call from a remote destination, this timer represents the amount of time where the user still has the option of resuming the call from a deskphone.

d)  In the **Remote Destination Limit** field, enter the number of remote destinations that a user is permitted to have for single number reach (SNR) targets.

**Step 5**  Complete the remaining fields in the **End User Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 6**  Click **Save**.
Configure Mobility Users through Bulk Administration

Use this procedure to use Bulk Administration's **Update Users** menu to add the Mobility feature to existing end users by bulk.

**Note**

Bulk Administration contains other features that allow you to update existing users by bulk. For example, you can use the Export and Import functions to import a CSV file with the new Mobility settings. For more information, see the *Bulk Administration Guide for Cisco Unified Communications Manager*.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Bulk Administration > Users > Update Users > Query**.

**Step 2** Apply the filter and click **Find** to select the users whom you want to assign as mobility users.

**Step 3** Click **Next**.

**Step 4** In the **Mobility Information** section, modify the following four fields by first checking the check box on the far left to indicate that this field is to be updated, and then configuring the setting on the right as follows:

- **Enable Mobility**—Check this check box to enable the users provisioned with this template for Mobility features.
- **Enable Mobile Voice Access**—Check this check box for provisioned users to be able to use Mobile Voice Access.
- **Maximum Wait Time for Desk Pickup**—This field represents the amount of time, after hanging up a call on a mobile phone, that you have to resume the call on your deskphone.
- **Remote Destination Limit**—This field represents the number of Remote Destinations and Mobile Identities that you can assign to users whom are provisioned through this template.

**Step 5** Under **Job Information**, check **Run Immediately**.

**Step 6** Click **Submit**.

Provision Mobility Users Through LDAP

If you have not yet synced your LDAP directory, you can use this procedure to configure synced end users with mobility capability through the Feature Group Template configuration. Newly synced users inherit the mobility settings from the template.

**Note**

This method works only if you have not yet synced your LDAP directory. You cannot assign new feature group template configurations to an LDAP directory sync after the initial sync has occurred.
**Configure Mobility for IP Phones**

Complete these tasks to configure mobility features for Cisco IP Phones. This includes setting up Single Number Reach (SNR) and the Move To Mobile feature. This provides users with a single enterprise number that rings all their devices, in addition to an enterprise-level voicemail that can be reached no matter which device rings. And also, users are able to transfer active calls between their deskphone and mobile device.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Softkey Template for Mobility, on page 16</td>
<td>Configures a mobility softkey template for Cisco IP Phones that includes the Mobility softkey. Users can transfer calls from their deskphone to a mobile phone by pressing the softkey.</td>
</tr>
</tbody>
</table>
Configure Softkey Template for Mobility

Use this procedure to configure a softkey template that includes the Mobility softkey. The softkey will be enabled for all phones that use this template.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Device Settings > Softkey Template. The Softkey Template Configuration window appears.

Step 2  To create a new softkey template do the following. Otherwise, proceed to the next step.
   a) Click Add New.
   b) Select a default template and click Copy.
   c) In the Softkey Template Name field, enter a new name for the template.
   d) Click Save.

Step 3  To add mobility softkeys to an existing template.
   a) Enter search criteria and click Find.
   b) Choose an existing template.

Step 4  (Optional) Check the Default Softkey Template check box if you want 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  Click Save.
Step 6 From the Related Links drop-down list, choose Configure Softkey Layout and click Go.

Step 7 From the Select a Call State to Configure drop-down list, choose the call state for which you want to add the softkey. Typically, you will want to add the softkey for both the OnHook and Connected call states.

Step 8 From the Unselected Softkeys list, choose the Mobility softkey and use the arrows to move the softkey to the Selected Softkeys list. Use the up and down arrows to change the position of the new softkey.

Step 9 To display the softkey in additional call states, repeat the previous step.

Step 10 Click Save.

Note If you created a new softkey template, you can assign the template to a phone through the Phone Configuration window or to a group of phones through Bulk Administration's Update Phones menu.

There are several methods to assign softkey template to phones during provisioning. For example, you can use the Universal Device Template configuration, or you can assign it as the default device profile for a specific model.

---

**Enable Mobility within Feature Control Policy**

If you have configured feature control policies to enable or disable features for Cisco IP Phones, then you will also have to enable Mobility within the policy that is used by your Cisco IP Phones. If the feature is disabled within the feature control policy configuration that is used by your phones, then the Mobility softkey will be disabled for all Cisco IP Phones that use that policy.

**Procedure**

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Feature Control Policy.

Step 2 Click Find and choose the applicable policy.

Note You can also choose Add New if you want to create a new feature control policy that you will assign to your phones to enable mobility, along with other associated features. You can assign the policy to phones through the Phone Configuration window, or to a set of phones through the Common Phone Profile Configuration. You can also assign the policy to a universal device template to assign the policy to phones as they are provisioned.

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

Step 4 In the Description field, enter a brief description for the feature control policy. The description can include up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).

Step 5 In the Feature Control Section, check both the Override Default check box and the Enable Setting check box that correspode to the Mobility softkey.

Step 6 Click Save.
Configure IP Phone for Mobility

If you have Single Number Reach or Move to Mobility configured, use this procedure to configure your deskphone with the Mobility feature so that enterprise calls can be redirected to a remote destination.

**Procedure**

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

**Step 2** Perform one of the following tasks:

- Click **Find** and select an existing phone to modify the settings.
- Click **Add New** and choose a phone from the **Phone type** drop-down list to add a new phone.

**Step 3** Click **Next**.

**Step 4** From the **SoftKey Template** drop-down list, choose the mobility softkey template that you configured.

**Step 5** From the **Owner User ID** drop-down list, choose the user account on which you enabled mobility.

**Step 6** (Optional) If you are using a **Feature Control Policy** to enable features, select the policy from the drop-down list.

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

Configure a Remote Destination Profile

Configures common settings that you want to apply to all the remote destination numbers for a user.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Device Profile > Remote Destination Profile**.

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

**Step 3** Enter a **Name** for the profile.

**Step 4** From the **User ID** drop-down list, select the end user to whom this profile applies.

**Step 5** From the **Device Pool** drop-down, select the device pool where this profile should reside.

**Step 6** Configure the remaining fields in the **Remote Destination Profile Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

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

**Step 8** Under **Association Information**, click **Add a New DN**.

**Step 9** In the **Directory Number** field, add the directory number of the user's desk phone.

Configure a Remote Destination

A remote destination is a virtual device that represents a mobile device where the user can be reached (for example, a home office phone, a mobile phone on a cellular network, or a PSTN phone). The remote destination carries many of the same settings as the user's desk phone.
1. When an enterprise user initiates a call from a remote destination to Cisco Jabber, Unified Communications Manager tries to establish a data call with Cisco Jabber by sending an INVITE message to Cisco TelePresence Video Communication Server (VCS). The call is established regardless of receiving a response from VCS.

2. If you have Self-Provisioning enabled, your end users can provision their own phones from the Self-Care Portal. Refer to the System Configuration Guide for Cisco Unified Communications Manager and the "Configure Self-Provisioning" chapter for details on configuring the system for self-provisioning and the "Provisioning End Users" part for details on enabling self-provisioning for users as a part of a User Profile.

### Procedure

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

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

**Step 3** In the **Destination** field, enter the number of the remote destination. For example, this could be a cellular number or PSTN number.

**Step 4** From the **Mobility User ID** field, select the mobility-enabled end user who uses this remote destination.

**Step 5** Check the **Enable Unified Mobility** features check box.

**Step 6** From the **Remote Destination Profile** drop-down, select the profile that you set up for the user who owns this remote destination.

**Step 7** From the **Single Number Reach Voicemail Policy** drop-down, configure the voicemail policy.

  a) From the **Single Number Reach Voicemail Policy** drop-down, configure the voicemail policy.

  b) Check the **Enable Single Number Reach** checkbox.

**Step 8** Check the **Move to Mobile** check box to include this remote destination to the list of available destinations when the user hits the **Mobility** softkey on their desk phone.

**Step 9** (Optional) Configure a **Ring Schedule** if you want to limit enterprise calls to this remote destination to just specific days such as office hours.

**Step 10** In the **When receiving a call during the above ring schedule** area, apply the list configured for this remote destination.

**Step 11** Configure the remaining fields on the **Remote Destination Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 12** Click **Save**.

### Configure an Access List

An access list is an optional remote destination configuration if you want to control which calls can ring which remote destinations, and at which times of day. The access list filters callers based on the Caller ID and can either allow calls or block calls during that remote destination’s ring schedule.
Phone users can configure their own access lists through the Self-Care Portal.

Procedure

**Step 1** From Cisco Unified CM Administration, choose **Call Routing > Class of Control > Access List**.

**Step 2** Click **Add New** to create an access list.

**Step 3** Enter a name and description to identify the new access list.

**Step 4** Associate the access list to a user by choosing an ID from the **Owner** drop-down list.

**Step 5** Choose one of the following options:

- **Allowed** — All numbers in the access list are allowed.
- **Blocked** — All numbers in the access list are blocked.

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

**Step 7** From the **Filter Mask** drop-down list, choose the filters that you want to apply to the access list:

- **Not Available** — All callers that advertise a not available status are added to the access list.
- **Private** — All callers that advertise a private status are added to the access list.
- **Directory Number** — All directory numbers or directory strings that you specify are added to the access list. If you choose this option, add a number or number string in the **DN Mask** field.

**Step 8** Choose **Save**.

**Step 9** Apply the access list to a remote destination:

a) From Cisco Unified CM Administration, choose **Device > Remote Destination** and reopen the remote destination that you created.

b) Configure the **Ring Schedule** for this access list and do either of the following:

- If you created an allowed access list, click the **Ring this destination only if caller is in** radio button and choose the access list that you created from the drop-down list.

- If you created a blocked access list, click the **Do not ring this destination if caller is in** radio button and choose the access list that you created from the drop-down list.

c) Click **Save**.

---

**Configure Mobile Voice Access**

Complete the following tasks to configure the system for Mobile Voice Access, which lets users place enterprise-anchored calls from any device. Users dial a system IVR for authentication, following which the call is sent out as an enterprise call that will appear to the end user as if the call were sent from the office phone.

**Before you begin**

To use Mobile Voice Access:
- Users must be enabled as mobility users with the **Enable Mobile Voice Access** option checked within **End User Configuration** For details, see **Configure a Mobility User, on page 13.**

- Interactive Voice Response service must be active, and included in a Media Resource Group List that the trunk uses.

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Acticate the Cisco Unified Mobile Voice Access Service, on page 22</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Enable Mobile Voice Access, on page 22</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Directory Number for Mobile Voice Access, on page 23</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Restart Cisco CallManager Service, on page 23</td>
</tr>
</tbody>
</table>
| **Step 5** | Configure a gateway for legacy MVA or enterprise feature access (EFA) by performing one of the following tasks:  
  - Configure an Existing H.323 or SIP Gateway for Remote Access, on page 24  
  - Configure a New H.323 Gateway for Remote Access, on page 25 |

**Note** Gateway configuration is no longer mandatory for Mobile Voice Access. This is an optional configuration only if you want to configure legacy Mobile Voice Access through an ISR G2 router.

Depending on your system requirements, you can add a new gateway or configure an existing gateway to handle calls that come from outside the enterprise through MVA or EFA.

If you have an existing H.323 or SIP PSTN gateway in your system, you can configure it for MVA. This function is accessed by calling a system-configured DID number that is answered and handled by an H.323 or SIP VoiceXML (VXML) gateway. After you configure your gateway, it uses a vxml script on the publisher node to pull the interactive voice response (IVR) prompts that are played to the MVA users. These prompts request user authentication and input of a number that users must dial on their phone keypad.

If you do not have an existing H.323 or SIP PSTN gateway and you want to configure mobile voice access, you must add a new H.323 gateway and configure it for MVA through the gateway configuration task.
### Activate the Cisco Unified Mobile Voice Access Service

Activate this service on your publisher node.

**Procedure**

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

**Step 2** From the **Server** drop-down list, choose the publisher node.

**Step 3** Click **Go**.

**Step 4** Under **CM Services**, check the **Cisco Unified Mobile Voice Access Service** check box.

**Step 5** Click **Save**.

### Enable Mobile Voice Access

Configure service parameters to enable Mobile Voice Access (MVA) and to specify the directory number or PSTN DID number that users can dial in order to reach the IVR.

**Before you begin**

The Cisco Unified Mobile Voice Access feature service must be activated for Mobile Voice Access to work.

**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** Configure the following service parameters:

- **Enable Mobile Voice Access**—Set this parameter to **True**.
- **Mobile Voice Access Number**—Enter the access number that you want users to dial when they access the enterprise.

**Step 5** Click **Save**.
Configure Directory Number for Mobile Voice Access

Configure mobile voice access (MVA) to assign sets of localized prompts for users who dial in from outside the enterprise.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose Media Resources > Mobile Voice Access.

**Step 2**
In the Mobile Voice Access Directory Number, enter the internal directory number (DN) to receive Mobile Voice Access calls from the gateway.

Enter a value between 1-24 digits in length. Valid values are 0-9.

**Step 3**
In the Localization pane, use the arrows to move the locales that you want to select to or from this pane.

**Note** Mobile Voice Access uses the first locale that appears in the Selected Locales pane in the Mobile Voice Access window. For example, if English United States appears first in the Selected Locales pane, the Cisco Unified Mobility user hears English when the IVR is used during a call.

**Step 4**
Click Save.

What to do next

Restart Cisco CallManager Service, on page 23

**Restart Cisco CallManager Service**

After you enable the Mobile Voice Access feature, restart the Cisco CallManager service.

Procedure

**Step 1**
From Cisco Unified Serviceability, choose Tools > Control Center - Feature Services

**Step 2**
From the Server drop-down, select the Cisco Unified Communications Manager publisher node.

**Step 3**
Under CM Services, select the radio button that corresponds to the Cisco CallManager service.

**Step 4**
Click Restart.

What to do next

You have now completed all the tasks that are required to configure Unified Communications Manager with native Mobile Voice Access support. However, if you want to configure legacy Mobile Voice Access where an ISR G2 router provides the IVR and voice prompts, you can complete either of the following two optional tasks:

- Configure an Existing H.323 or SIP Gateway for Remote Access, on page 24
- Configure a New H.323 Gateway for Remote Access, on page 25
Configure an Existing H.323 or SIP Gateway for Remote Access

If you have an existing H.323 or SIP PSTN gateway in your system, you can configure it for MVA. This function is accessed by calling a system-configured DID number that is answered and handled by an H.323 or SIP VoiceXML (VXML) gateway. After you configure your gateway, it uses a vxml script on the publisher node to pull the interactive voice response (IVR) prompts that are played to the MVA users. These prompts request user authentication and input of a number that users must dial on their phone keypad.

Before you begin

Configure Directory Number for Mobile Voice Access, on page 23

Procedure

Step 1 Configure the T1/E1 controller for PRI from the PSTN.

Example:

controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-24

Step 2 Configure the serial interface for the PRI (T1/E1).

Example:

interface Serial 1/0:23
ip address none
logging event link-status none
isdn switch-type primary 4ess
isdn incoming-voice voice
isdn bchan-number-order ascending
no cdp enable

Step 3 Load the VXML application from the publisher node.

Example:

Sample configuration for IOS Version 12.3 (13) and later:

application service CCM
http://<Unified CM Publisher IP Addr>:8080/ccmivr/pages/IVRMainpage.vxml

Example:

Sample configuration before IOS Version 12.3(12):

call application voice Unified CCM
http://<Unified CM Publisher IP Addr>:8080/ccmivr/pages/IVRMainpage.vxml

Caution Although VXML was added in Version 12.2(11), other versions such as 12.3(8), 12.3(9), 12.3(14)T1, and 12.2(15) have VXML issues.

Step 4 Configure the dial peer to associate the Cisco Unified Mobility application with system remote access.

Example:

Sample configuration for IOS 12.3(13) and later:
Sample configuration for iOS 12.3(12) and earlier:

dial-peer voice 100 pots
  application CCM (Cisco Unified Mobility VXML application)
  incoming called-number 58888

(58888 represents the mobile voice access (MVA) number)

Step 5
Add a dial peer to transfer the calls to the MVA DN.

Example:
Sample configuration for primary Unified Communications Manager:

dial-peer voice 101 voip
  preference 1
  destination-pattern <Mobile Voice Access DN>
  session target ipv4:10.1.30.3
  codec g711ulaw
  dtmf-relay h245-alphanumeric
  no vad

Example:
Sample configuration for secondary Unified Communications Manager (if needed):

dial-peer voice 102 voip
  preference 2
  destination-pattern <Mobile Voice Access DN>
  session target ipv4:10.1.30.4
  codec g711ulaw
  dtmf-relay h245-alphanumeric
  no vad

Note If a generic dial peer is already configured to terminate the calls and is consistent with the MVA DN, you do not need to perform this step.

Example:
Sample configuration for SIP gateway VoIP dial-peer:

dial-peer voice 80 voip
  destination-pattern <Mobile Voice Access DN>
  rtp payload-type nse 99
  session protocol sipv2
  session target ipv4:10.194.107.80
  incoming called-number .T
  dtmf-relay rtp-nte
  codec g711ulaw

---

**Configure a New H.323 Gateway for Remote Access**

If you do not have an existing H.323 or SIP PSTN gateway and you want to configure mobile voice access, you must add a new H.323 gateway and configure it for MVA functionality by using the hairpinning method. From a technical standpoint, this method refers to using a second gateway to receive the inbound call, apply
the MVA service and then the inbound call leg returns to the PSTN gateway (original source) after the system applies the MVA service.

---

**Note**

If you use Mobile Voice Access with hairpinning, users calling into your system will not be identified automatically by their caller ID. Instead, users must enter their remote destination number manually before they enter their PIN. The reason is that the PSTN gateway must first route the call to Unified Communications Manager to reach the hairpinned Mobile Voice Access gateway. Because of this route path, the conversion of the calling number from a mobile number to an enterprise directory number occurs before the Mobile Voice Access gateway handles the call. As a result, the gateway is unable to match the calling number with a configured remote destination, and therefore the system prompts users to enter their remote destination number.

---

**Before you begin**

*Configure Directory Number for Mobile Voice Access, on page 23*

**Procedure**

**Step 1**

Load the VXML application from the publisher node.

**Example:**

Sample configuration for IOS Version 12.3 (13) and later:

```
application service CCM
http://<Unified CM Publisher IP Addr>:8080/ccivr/pages/IVRMainpage.vxml
```

**Example:**

Sample configuration before IOS Version 12.3(12):

```
call application voice CCM
http://<Unified CM Publisher IP Addr>:8080/ccivr/pages/IVRMainpage.vxml
```

**Caution** Although VXML was added in Version 12.2(11), other versions such as 12.3(8), 12.3(9), 12.3(14)T1, and 12.2(15) have VXML issues.

**Step 2**

Configure the dial-peer to associate the Cisco Unified Mobility application with system remote access.

**Example:**

Sample configuration for IOS 12.3(13) and later:

```
dial-peer voice 1234567 voip
service CCM
incoming called-number 1234567
codec g711u
session target ipv4:<ip_address of call manager>
```

**Example:**

Sample configuration for IOS 12.3(12) and earlier:

```
dial-peer voice 1234567 voip
application CCM
incoming called-number 1234567
codec g711u
session target ipv4:<ip_address of call manager>
```
Step 3  Add a dial-peer for transferring calls to the Mobile Voice Access (MVA) DN.

Example:
Sample configuration for primary Unified Communications Manager:

dial-peer voice 101 voip
preference 1
destination-pattern <Mobile Voice Access DN>
session target ipv4:10.1.30.3
voice-class h323 1
codec g711ulaw
dtmf-relay h245-alphanumeric
novad

Example:
Sample configuration for secondary Unified Communications Manager (if needed):

dial-peer voice 102 voip
preference 2
destination-pattern <Mobile Voice Access DN>
session target ipv4:10.1.30.4
voice-class h323 1
codec g711ulaw
dtmf-relay h245-alphanumeric
novad

Note  If a generic dial peer is already configured to terminate the calls and is consistent with the MVA DN, you do not need to perform this step.

Step 4  Configure hairpin.

voice service voip
allow-connections h323 to h323

Step 5  On the Unified Communications Manager, create a new route pattern to redirect the incoming MVA number to the H.323 gateway that has the vxml script loaded. Ensure that the incoming CSS of the gateway can access the partition in which the new route pattern is created.

Configure Enterprise Feature Access

You can use Enterprise Feature Access from a remote destination for:

- Two-stage dialing to place enterprise calls from a configured remote destination. Calls appear to the called party as if they were placed from an associated desk phone.

- Remote destination access to mid-call features through EFA codes that are sent using DTMF digits sent from the remote destination.

Note  Unlike Mobile Voice Access, with Enterprise Feature Access you must be calling from a configured remote destination.
**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Call Routing > Mobility > Enterprise Feature Access Number Configuration**.

**Step 2** In the **Number** field, enter the unique DID number that mobile users will dial from a remote destination in order to access the Enterprise Feature Access feature.

**Step 3** From the **Route Partition** drop-down list, choose the partition where the DID resides.

**Step 4** (Optional) Check the **Default Enterprise Feature Access Number** check box to make this EFA number the default for this system.

**Step 5** Click **Save**.

**Step 6** Configure the Enterprise Feature Access service parameters:

a) From Cisco Unified CM Administration, choose **System > Service Parameters**.

b) From the **Server** drop-down list, choose the publisher node.

c) From the **Service** drop-down list, choose **Cisco CallManager**.

d) Set the **Enable Enterprise Feature Access** service parameter to **True**.

e) (Optional) In the **Clusterwide Parameters (System - Mobility)** area, edit the DTMF digits that you must enter to access midcall features through Enterprise Feature Access. For example, you could edit the **Enterprise Feature Access Code for Hold** service parameter, which has a default value of *81. The default values are as follows:

- Hold: *81
- Exclusive Hold: *82
- Resume: *83
- Transfer: *84
- Conference: *85
- Session Handoff: *74
- Starting Selective Recording: *86
- Stopping Selective Recording: *87
- Hunt group login—enter a new code
- Hunt group logout—enter a new code

e) Click **Save**.

---

**Configure Intelligent Session Control**

Configure the system so that inbound calls to a remote destination are rerouted to an associated enterprise number, if one is available. This provides automatic call anchoring within the enterprise for mobility calls, providing cost savings and added Unified Communications functionality.
Configure Mobility Service Parameters

Use this procedure to configure optional Mobility-related service parameters.

Procedures

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 Configure any service parameters that you want to edit. The Mobility-related parameters are listed under the following headings. For help descriptions, click the parameter name:

- **Clusterwide Parameters (System - Mobility)**
- **Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)**
- **Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)**

Step 5 Click Save.

Configure Cisco Jabber Dual-Mode

Complete these tasks to configure Cisco Jabber on iPhone or Android as dual-mode mobile devices that can connect over WiFi. Cisco Jabber registers to Unified Communications Manager over WiFi and can be reached through an enterprise number if Single Number Reach is enabled in the user's mobile identity.
Procedure

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<th>Command or Action</th>
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<td>Configure a mobility profile to send consistent caller ID to Jabber mobile clients that are placing Dial through Office calls.</td>
</tr>
<tr>
<td>Step 2 Add a Dual-Mode Device for Cisco Jabber, on page 31</td>
<td>Configure a dual-mode device type for Cisco Jabber on iPhone or Android clients.</td>
</tr>
<tr>
<td>Step 3 Configure a Mobility Identity, on page 34</td>
<td>Add a Mobility Identity to the Jabber mobile client that points to the device phone number (that is, the iPhone number) to provide calling when Jabber roams out of WiFi range. Enable Single Number Reach destination for the Mobile Identity.</td>
</tr>
<tr>
<td>Step 4 Required: Configure Handoff Number, on page 34</td>
<td>Configure a handoff number for dual-mode devices that are leaving enterprise. Even when the device disconnects from the enterprise WiFi network the call can be maintained without interruption by reconnecting to a remote mobile or cellular network.</td>
</tr>
</tbody>
</table>

Configure Other Dual-Mode Devices

Complete these tasks to configure other dual-mode mobile devices that can place calls over the cellular network and can also connect over WiFi. For example:

- Carrier-Integrated Mobile Devices that connect over Fixed Mobile Convergence (FMC) networks.
- IMS-integrated Mobile Devices over IP Multimedia networks

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
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</thead>
<tbody>
<tr>
<td>Step 1 Add a Dual-Mode Device for Cisco Jabber, on page 31</td>
<td>Configure an IMS or FMC dual-mode device.</td>
</tr>
<tr>
<td>Step 2 Configure a Mobility Identity, on page 34</td>
<td>Add a Mobility Identity that points to the phone number of the actual device.</td>
</tr>
<tr>
<td>Step 3 Required: Configure Handoff Number, on page 34</td>
<td>Configure a handoff number for dual-mode devices that are leaving the enterprise. Even when the device disconnects from the enterprise WiFi network the call can be maintained without interruption by reconnecting to a remote mobile or cellular network.</td>
</tr>
</tbody>
</table>
Configure a Mobility Profile

Configure a mobility profile for dual-mode Cisco Jabber on iPhone and Android clients. The profile configures the client with a consistent caller ID for dial via office calls.

From a technical standpoint, this caller ID is sent during the dial via office reverse (DVO-R) callback portion of a call to the mobility identity or alternate callback number. DVO-R call feature uses enbloc dialing. If no mobility profile is assigned to the mobility identity or if the Callback Caller ID field is left blank, the system sends the default enterprise feature access number.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Mobility > Mobility Profile.
Step 2 Click Add New.
Step 3 Enter a Name for the profile.
Step 4 From the Mobile Client Calling Option drop-down list, select Dial via Office Reverse.
   Note Despite the field options, Dial via Office Forward is not available.
Step 5 Configure a Callback Caller ID for Dial-via-Office Reverse.
Step 6 Configure the fields in the Mobility Profile Configuration window. For more information on the fields and their configuration options, see the system Online Help.
Step 7 Click Save.

Add a Dual-Mode Device for Cisco Jabber

Configure a dual-mode device type for Cisco Jabber on iPhone or Android clients.

Before you begin

Make sure that your end users are mobility-enabled. Also, if you want to add remote destinations to your Jabber client, make sure that you have a softkey template that includes the Mobility softkey.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Perform one of the following:
   • Click Find to edit an existing device.
   • Click Add New and select either Cisco Dual Mode for Android or Cisco Dual Mode for iPhone as the phone model, to add a new device. Click Next.
Step 3 Configure the fields in the Phone Configuration window. For more information on the fields and their configuration options, see the Related Topics section.
For detailed information about product specific configuration layout fields, see your Jabber client documentation at [http://www.cisco.com/go/jabber](http://www.cisco.com/go/jabber).

**Step 4** Configure the following mandatory fields:

- Device Name
- Device Pool
- Softkey Template
- Owner User ID—The user must have mobility enabled.
- Mobility User ID—The user must have mobility enabled.
- Device Security Profile
- SIP Profile

**Step 5** Click **Save**.

**Step 6** Add a directory number:

a) In the left Association area, click **Add a New DN**.

b) Enter a new **Directory Number** and click **Save**.

c) Complete any fields that you want in the **Directory Number Configuration** window and click **Save**. For more information on the fields and their configuration options, see the system Online Help.

d) Click **Associate End Users**.

e) Click **Find** and select the mobility-enabled end user whom owns this DN.

f) Click **Add Selected**.

g) Click **Save**.

---

**What to do next**

Add a Mobility Identity that points to the phone number of the iPhone or Android device. This allows you to transfer the call to the phone if you move out of Wi-Fi range. You can also add the device as a Single Number Reach destination. For details, see the system Online Help. Configure a Mobility Identity, on page 34.

Optionally, add Remote Destinations and Single Number Reach to your Cisco Jabber client. When someone calls the Jabber client, the remote destination rings as well. Configure a Remote Destination, on page 18.

---

**Dual-Mode Device Configuration Fields**

*Table 1: Dual-Mode Device Configuration Fields*

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Softkey Template</td>
<td>Choose the Mobility Softkey template.</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>Choose the user ID of the assigned phone user. The user ID is recorded in the</td>
</tr>
<tr>
<td></td>
<td>call detail record (CDR) for all calls made from this device.</td>
</tr>
<tr>
<td>Mobility User ID</td>
<td>Choose the user ID of the person to whom this dual-mode phone is assigned.</td>
</tr>
</tbody>
</table>
Choosethesecurityprofiletoapplytothedevice.
YoumustapplyasecurityprofiletoallphonesthatareconfiguredinCisco
UnifiedCommunicationsManagerAdministration.Toenablesecurityfeatures
foraphone,youmustconfigureanewsecurityprofileforthedevicetypeand
protocol, and then apply it to the phone.

Chooseacallingsearchspaceforroutingcallstoconfiguredremotedestinations
andmobilityidentitiesthatareconfiguredforthisdevice.

Choose Standard SIP Profile for Mobile Device.

Add Other Dual-Mode Device

Use this procedure to add another dual-mode device (for example, a Carrier-integrated Mobile Device for
network-based FMC, or an IMS-integrated Mobile Device.

Before you begin

Make sure that your end users are mobility-enabled. Refer to topics earlier in this chapter for details on how
to enable mobility for users.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
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<tr>
<td>1</td>
<td>From Cisco Unified CM Administration, choose Device &gt; Phone.</td>
</tr>
<tr>
<td>2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>3</td>
<td>From the Phone Model drop-down list Carrier-integrated Mobile Device or IMS-integrated Mobile Device.</td>
</tr>
<tr>
<td>4</td>
<td>Configure the following mandatory fields:</td>
</tr>
<tr>
<td></td>
<td>• Device Name</td>
</tr>
<tr>
<td></td>
<td>• Device Pool</td>
</tr>
<tr>
<td></td>
<td>• Owner User ID—The user must have mobility enabled.</td>
</tr>
<tr>
<td></td>
<td>• Mobility User ID—The user must have mobility enabled.</td>
</tr>
</tbody>
</table>
| 5    | Configure the remaining fields in the Phone Configuration window. For more information on the fields and
their configuration options, see the system Online Help. |
| 6    | Click Save. |
| 7    | Add a directory number: |
|      | a) In the left Association area, click Add a New DN. |
|      | b) Enter a new Directory Number and click Save. |
|      | c) Complete any fields that you want in the Directory Number Configuration window and click Save. For
more information on the fields and their configuration options, see the system Online Help. |
|      | d) Click Associate End Users. |
|      | e) Click Find and select the mobility-enabled end user whom owns this DN. |
|      | f) Click Add Selected. |
g) Click Save.

**Configure a Mobility Identity**

Add a Mobility Identity that points to the phone number of the device if you want to enable the device as a Single Number Reach that can be reached through the enterprise number.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Phone</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Enter search criteria if needed, click <strong>Find</strong>, and choose the dual-mode device that you created.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click <strong>Add New Mobility Identity</strong>.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Destination</strong> field, enter the phone number of the mobile device. For example, for a Cisco Jabber on iPhone client, this would be the phone number of the iPhone.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Cisco Jabber only. Select the <strong>Mobility Profile</strong> that you configured.</td>
</tr>
<tr>
<td>Step 6</td>
<td>If you want to make this Mobile Identity available from an enterprise phone number:</td>
</tr>
<tr>
<td></td>
<td><strong>a)</strong> Check the <strong>Enable Single Number Reach</strong> check box.</td>
</tr>
<tr>
<td></td>
<td><strong>b)</strong> Configure a <strong>Single Number Reach Voicemail</strong> policy.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Configure a <strong>Dial-via-Office Reverse Voicemail</strong> policy.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Configure the fields on the <strong>Mobility Identity Configuration</strong> window. For more information on the fields and their configuration options, see the system Online Help.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**Note** If you want to apply a Ring Schedule and access list to limit calls to this mobile identity to specific times and users, **Configure an Access List**, on page 19.

**Configure Handoff Number**

Configure handoff mobility for dual-mode phones if you want your system to preserve a call while the user moves out of the enterprise. Even when a user's device disconnects from the enterprise WiFi network and reconnects to the mobile voice or cellular network, an in-progress call is maintained without interruption.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Mobility &gt; Handoff Configuration</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the <strong>Handoff Number</strong> field, enter the direct inward dialing (DID) number for handoff between the Wi-Fi and mobile voice or cellular network.</td>
</tr>
<tr>
<td></td>
<td>For numbers that start with the international escape character (+), you must precede the + with a backslash (). Example: \+15551234.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Route Partition</strong> drop-down list, choose the partition to which the handoff DID number belongs.</td>
</tr>
</tbody>
</table>
**Cisco Unified Mobility Call Flow**

This section describes the incoming and outgoing call flows of Cisco Unified Mobility commonly known as Single Number Reach (SNR). Unified Communications Manager supports the separate calling party number and billing number feature when SNR is configured for users to allow desk phones to extend calls to mobile devices.

For example, User-A calls from a PSTN network to User-B whose directory number configured to SNR. If **Enable External Presentation Name and Number** check box is checked in SIP profile and **Display External Presentation Name and Number** service parameter value set to True, then Unified Communications Manager displays the FROM header information on both the User-B’s desk phone and the configured remote destination device. In the same way, if any one option is disabled, Unified Communications Manager displays P-Asserted-Identity (PAID) header information on the called device.

Similarly, in outgoing call scenario User B (SNRD line) configured with External Presentation Information on Directory Number configuration page initiates a call to a PSTN network through a SIP trunk. If **Enable External Presentation Name and Number** is configured in its SIP profile, then, Unified Communications Manager send the External Presentation Information in the FROM header of the outgoing SIP message to display on the called device.

If **Enable External Presentation Name and Number** check box is disabled, then Unified Communications Manager sends the directory number information in the FROM and PAID to display on the called device and configured External Presentation Information in the X-Cisco-Presentation header.

If you check the **Anonymous External Presentation** check box, the configured **External Presentation Name** and **External Presentation Number** are removed from the respective fields and external presentation displayed as anonymous on the called device.

For more details on Configuring External Presentation Information, see *Configure Directory Number* chapter in the System Configuration Guide for Cisco Unified Communications Manager.

**Cisco Unified Mobility Interactions**

*Table 2: Cisco Unified Mobility Interactions*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Call Pickup</td>
<td>Cisco Unified Mobility interacts with auto call pickup depending on how you configured the service parameter. When the <strong>Auto Call Pickup Enabled</strong> service parameter is set to True, users must press only the <strong>PickUp</strong> softkey to pick up a call. If the service parameter is set to False, users must press the <strong>PickUp</strong>, <strong>GPickUp</strong>, or <strong>OPickUp</strong> softkey and then the Answer softkey.</td>
</tr>
<tr>
<td>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>---------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Automatic Alternate Routing</strong></td>
<td>Cisco Unified Mobility supports automatic alternate routing (AAR) as follows:</td>
</tr>
<tr>
<td></td>
<td>• If a rejection occurs because of a lack of bandwidth for the location-based service, the rejection triggers AAR and reroutes the call through the PSTN so the caller does not need to hang up and redial.</td>
</tr>
<tr>
<td></td>
<td>• If a rejection occurs because of resource reservation protocol (RSVP), however, AAR is not triggered for calls to remote destinations and the call stops.</td>
</tr>
<tr>
<td><strong>Extend and Connect</strong></td>
<td>Users who need the capabilities of both Cisco Unified Mobility and Extend and Connect can configure the same remote destination on the remote device profile and CTI remote device types when the owner ID of both device types is the same. This configuration allows Cisco Unified Mobility features to be used concurrently with Extend and Connect.</td>
</tr>
<tr>
<td></td>
<td>For more information, see the “Extend and Connect” chapter.</td>
</tr>
<tr>
<td><strong>External Call Control</strong></td>
<td>If external call control is configured, Unified Communications Manager follows the route decision from the adjunct route server for these Cisco Unified Mobility features:</td>
</tr>
<tr>
<td></td>
<td>• Cisco Unified Mobility</td>
</tr>
<tr>
<td></td>
<td>• Mobile voice access</td>
</tr>
<tr>
<td></td>
<td>• Enterprise feature access</td>
</tr>
<tr>
<td></td>
<td>• Dial via office</td>
</tr>
<tr>
<td></td>
<td>Unified Communications Manager does not send a routing query for the following Cisco Unified Mobility features:</td>
</tr>
<tr>
<td></td>
<td>• Cell pickup</td>
</tr>
<tr>
<td></td>
<td>• Desk pickup</td>
</tr>
<tr>
<td></td>
<td>• Session handoff</td>
</tr>
<tr>
<td><strong>Intelligent Session Control and Session Handoff</strong></td>
<td>For direct calls to remote destinations that are anchored to the enterprise number, mobile users can use the session handoff feature to hand off the call to their deskphones.</td>
</tr>
<tr>
<td></td>
<td>You must enable Cisco Unified Mobility before you implement intelligent session control.</td>
</tr>
<tr>
<td><strong>Licensing</strong></td>
<td>Cisco Unified Mobility is included in all user-based licenses from basic to professional.</td>
</tr>
<tr>
<td><strong>Local Route Groups</strong></td>
<td>For single number reach calls to a remote destination, the device pool of the originating calling party determines the selection of the standard local route group.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> Local Route Group is not supported when the AgentGreeting with BiB (Built in Bridge) is invoked.</td>
</tr>
</tbody>
</table>
interaction feature

Each remote destination supports a maximum of six active calls. However, the number of supported calls depends on the Unified Communications Manager configuration.

For example, the Cisco Unified Mobility user receives a call while the user already has six calls for the remote destination or while the user is using DTMF to transfer or conference a call from the remote destination.

The received call is sent to the enterprise voice mail when:

- The number of calls with user exceeds Busy trigger configuration
- CFB is configured
- All shared lines are busy

**Note**

The calls sent to the enterprise voice mail is not based on the maximum supported calls.

SIP Trunks with Cisco Unified Border Element

Cisco Unified Mobility supports the Cisco Unified Mobility feature without midcall features over SIP trunks with Cisco Unified Border Element (CUBE).

### Cisco Unified Mobility Restrictions

**Table 3: Cisco Unified Mobility Interactions**

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Answer</td>
<td>A remote destination call does not work when auto answer is enabled.</td>
</tr>
<tr>
<td>Call Forwarding Unregistered</td>
<td>Call Forward Unregistered (CFUR) support for Cisco Jabber on iPhone and Android is as follows:</td>
</tr>
<tr>
<td></td>
<td>• CFUR is supported if Cisco Jabber on iPhone or Android does not have either a mobile identity or remote destination configured.</td>
</tr>
<tr>
<td></td>
<td>• CFUR is not supported, and will not work, if a Remote Destination is configured</td>
</tr>
<tr>
<td></td>
<td>• CFUR is not supported, and will not work, if a Mobile Identity is configured with a mobile phone number and Single Number Reach is enabled.</td>
</tr>
<tr>
<td></td>
<td>If you have a mobile identity or remote destination configured, use Call Forward Busy and Call Forward No Answer instead.</td>
</tr>
<tr>
<td>Call Queuing</td>
<td>Unified Communications Manager does not support call queuing with Cisco Unified Mobility.</td>
</tr>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>-------------</td>
</tr>
<tr>
<td>Conferencing</td>
<td>Users cannot initiate a meet-me conference as conference controller by using mobile voice access, but they can join a meet-me conference. If an existing conference call is initiated from a shared-line IP phone or dual-mode phone or smartphone that is a remote destination, no new conference party can be added to the existing conference after the call is sent to a mobile phone or a dual-mode handoff action occurs. To permit the addition of new conference parties, use the Advanced Ad Hoc Conference Enabled service parameter. See the Related Topic section for more information.</td>
</tr>
<tr>
<td>Dialing + Character from Mobile Phones</td>
<td>Users can dial a + sign through dual-tone multifrequency (DTMF) on a mobile phone to specify the international escape character. Cisco Unified Mobility does not support + dialing through DTMF for IVR to make an outgoing call from a mobile phone to an enterprise IP phone for which the directory number contains the + character. Cisco Unified Mobility does not support + dialing through DTMF for two-stage dialing to make an outgoing call from a mobile phone to an enterprise IP phone for which the directory number contains the + character.</td>
</tr>
<tr>
<td>Do Not Disturb on the Desk Phone and Direct Calls to Remote Destination</td>
<td>If do not disturb (DND) is enabled on a desk phone, the desk phone cannot be placed in the remote In use state and the call is not anchored in the following scenarios: • DND is enabled with the call reject option. • DND is activated by pressing the DND softkey on the desk phone. If DND is enabled with the ring off option, however, the call is anchored.</td>
</tr>
<tr>
<td>Dual-Mode Phones</td>
<td><strong>Dual-Mode Handoff and Caller ID</strong> The handoff DN method of dual-mode handoff requires a caller ID in the cellular network. The mobility softkey method does not require caller ID. <strong>Dual-Mode Phones and CTI Applications</strong> While a dual-mode phone is in Wi-Fi enterprise mode, no CTI applications control it or monitor it. The In Use Remote indicator for dual-mode phones on a shared line call in the WLAN disappears if the dual-mode phone goes out of WLAN range. <strong>Dual-Mode Phones and SIP Registration Period</strong> For dual-mode phones, Unified Communications Manager determines the registration period by using the value in the Timer Register Expires (seconds) field of the SIP profile that associates with the phone, not the value that the SIP Station KeepAlive Interval service parameter specifies. The standard SIP profile for mobile devices determines the registration period as defined by the Time Register Expires field in that profile.</td>
</tr>
</tbody>
</table>
### Enterprise Features From Cellular Networks

Enterprise features from cellular networks require out-of-band DTMF. When using intercluster DNAs as remote destinations for an IP phone over a SIP trunk (either intercluster trunk or gateway), check the **Require DTMF Reception** check box when configuring the IP phone. This allows DTMF digits to be received out of band, which is crucial for enterprise feature access midcall features.

### Gateways and Ports

Both H.323 and SIP VoIP gateways are supported for mobile voice access. Cisco Unified Mobility features are not supported for T1 CAS, FXO, FXS and BRI.

SNR (Single Number Reach) is not supported with MGCP (Media Gateway Controlled Protocol).

### Jabber Devices

When initially configured, Jabber devices count as registered devices. These devices increase the count of registered devices in a node, set by the **Maximum Number of Registered Devices** service parameter.

### Locales

Cisco Unified Mobility supports a maximum of nine locales. If more than nine locales are installed, they appear in the Available Locales pane, but you can only save up to nine locales in the Selected Locales pane.

If you attempt to configure more than nine locales for Cisco Unified Mobility, the following message appears: “Update failed. Check constraint (informix.cc_ivruserlocale_orderindex) failed.”

### Maximum Wait Timer for Desktop Call Pickup

If a user presses the *81 DTMF code from a remote destination (either a smartphone or any other phone) to put a call on hold, the user desk phone displays the **Resume** softkey. However, the desk phone does not apply a timer for Desktop Call Pickup. The **Resume** key continues to be displayed even after the timeout that is configured for the end user to pick up the call elapses and the call is not dropped.

Instead, users should hang up the call on the remote phone, which triggers the desk phone to apply the timer for desktop call pickup. (Use the **Maximum Wait Time for Desk Pickup** field on the **End User Configuration** window to change this setting.)

### Multilevel Precedence and Preemption

Cisco Unified Mobility does not work with multilevel precedence and preemption (MLPP). If a call is preempted with MLPP, Cisco Unified Mobility features are disabled for that call.

### Overlap Sending

Overlap sending patterns are not supported for the Intelligent Session Control feature.

### Q Signaling

Mobility does not support Q signaling (QSIG).

### QSIG Path Replacement

QSIG path replacement is not supported.
### Service Parameters

Enterprise feature access service parameters apply to standard phones and smartphones; however, smartphones generally use one-touch keys to send the appropriate codes. You must configure any smartphones that will be used with Cisco Unified Mobility to use either the default codes for enterprise feature access or the codes that are specified in the smartphone documentation.

### Session Handoff

The following limitations apply to the session handoff feature:

- Session handoff can take place only from mobile phone to desk phone. For session handoff from desk phone to mobile phone, the current remote destination pickup method specifies that you must use send call to mobile phone.
- Only audio call session handoff is supported.

### Single Number Reach with Hunt Groups

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 **Logged Into Hunt Group** check box must be checked within the **Phone Configuration** window for that device.

### SIP Trunks

The Cisco Unified Mobility feature is supported only for primary rate interface (PRI) public switched telephone network (PSTN) connections.

For SIP trunks, Cisco Unified Mobility is supported over IOS gateways or intercluster trunks.

### SIP URI and Direct Calls to Remote Destination

The Intelligent session control feature does not support direct URI dialing. Therefore, calls that are made to a SIP URI cannot be anchored to an enterprise number.

### Unified Communications Manager publisher dependent features

In a cluster environment, the publisher must be reachable in order to enable or disable Single Number Reach. Some features may not function if the publisher is not actively running.

Mobile voice access is not available when the publisher node is not reachable; IVR prompts for Mobile Voice Access are stored only on the publisher.

### Video Calls

Cisco Unified Mobility services do not extend to video calls. A video call that is received at the desk phone cannot be picked up on the mobile phone.

### Mobile Voice Access (MVA)

Cisco 4000 Series Integrated Services Routers do not support Voice XML (VXML). Hence, when these routers function as Unified Communications gateways with Cisco Unified Communications Manager, they do not support Mobile Voice Access (MVA) application.

### Related Topics

- Ad Hoc Conferencing Service Parameters, on page 208
Cisco Unified Mobility Troubleshooting

Cannot Resume Call on Desktop Phone

**Problem** When a remote destination (mobile phone) is not a smartphone and a call to this mobile phone is anchored through Cisco Unified Communications Manager, the user can hang up the mobile phone and expect to see a **Resume** softkey on the user desktop phone to resume the call. The user cannot resume this call on the user desktop phone.

**Possible Cause** If the calling party receives a busy, reorder, or disconnect tone when the mobile phone hangs up, the mobile phone provider probably did not disconnect the media. No disconnect signals came from the provider. To verify this possibility, let the calling party wait for 45 seconds. After this wait, the service provider will time out and send disconnect signals, at which time Cisco Unified Communications Manager can provide a **Resume** softkey to resume the call.

- Add the following command to the gateway:
  
  ```
  voice call disc-pi-off
  ```

- For the Cisco CallManager service, set the **Retain Media on Disconnect with PI for Active Call** service parameter to **False**.
Cannot Resume Call on Desktop Phone
Device Mobility Overview

Device mobility lets mobile users roam between sites, taking on the site-specific settings of the local site. When this feature is configured, Cisco Unified Communications Manager matches the IP address of a roaming device to IP subnets in the Device Mobility configuration to determine the physical location of the device so that an appropriate device pool can be assigned. The settings from this dynamically-assigned device pool override the settings in the Phone Configuration for that device and ensure that voice quality and allocation of resources are appropriate for the new phone location.

For roaming mobile devices, this feature provides a more efficient use of network resources:

- When a mobile user moves to another location, call admission control (CAC) can ensure video and audio quality with the appropriate bandwidth allocations for that location.
- When a mobile user makes a PSTN call, the phone is routed to the local gateway. Otherwise, PSTN calls would first be routed back to the home site over IP WAN connections, and then on to a PSTN gateway at the home site.
- When a mobile user calls the home location, Cisco Unified Communications Manager can assign the appropriate codec for the region.

Site-Specific Settings

For roaming devices, Cisco Unified Communications Manager overwrites the following device pool parameters from the device configuration with values from the dynamically assigned device pool:

- Date/Time Group
- Region
- Location
- Network Locale
When networks span geographic locations outside the United States, you can configure device mobility groups to allow phone users to use their configured dial plan no matter where they roam. When a device is roaming but remains in the same device mobility group, Cisco Unified Communications Manager also overwrites the following device pool parameters:

- AAR Group
- AAR Calling Search Space
- Device Calling Search Space

When the phone returns to its home location, the system disassociates the roaming device pool, downloads the configuration settings for home location, and resets the device. The device registers with the home location configuration settings.

**Note**
Cisco Unified Communications Manager always uses the Communications Manager Group setting from the phone record. The device always registers to its home location Cisco Unified Communications Manager server even when roaming. When a phone is roaming, only network location setting such as bandwidth allocation, media resource allocation, region configuration, and AAR group get changed.

**Configuration**
This feature needs to be enabled at both the system-level, and at the device level. At the system level, this feature uses the following components:

- **Physical Location**—The physical location of the device pool. During registration, the system matches the device registration location to a subnet in the Device Mobility Info in order to assign an appropriate device pool.

- **Device Pool**—Location-specific device settings such as media resources, regions, and SRST references. For roaming devices, the system assigns the device pool that matches that device’s physical location.

- **Device Mobility Group**—A logical group of sites with similar dialing patterns. For example, an enterprise with a worldwide network might set up groups that represent individual countries. The device mobility group setting determines whether the device is moved within the same geographical entity, primarily to allow users to keep their own dial plans.

- **Device Mobility Info**—This info contains the subnets that the system provides for roaming devices, and the device pools that the system can assign to roaming devices that register to one of those subnets.

At the device level, the feature must be turned on for devices to use this feature.
Device Pool Assignment

This section describes how Unified Communications Manager assigns device pools when device mobility is enabled. Depending on whether the device is roaming, the device may be assigned a device pool in the local site, or it may use the device pool from its home site.

Following initialization, the device mobility feature operates according to the following process:

1. A phone device record gets created for an IP phone that is provisioned to be mobile, and the phone gets assigned to a device pool. The phone registers with Unified Communications Manager, and an IP address gets assigned as part of the registration process.

2. Unified Communications Manager compares the IP address of the device to the subnets that are configured for device mobility in the Device Mobility Info Configuration window. The best match uses the largest number of bits in the IP subnet mask (longest match rule). For example, the IP address 9.9.8.2 matches the subnet 9.9.8.0/24 rather than the subnet 9.9.0.0/16.

3. If the device pool in the phone record matches the device pool in the matching subnet, the system considers the phone to be in its home location, and the phone retains the parameters of its home device pool.

4. If the device pool in the phone record does not match the device pools in the matching subnet, the system considers the phone to be roaming. The following table describes possible scenarios for device mobility and the system responses.
Table 4: Device Mobility Scenarios

<table>
<thead>
<tr>
<th>Scenario</th>
<th>System Response</th>
</tr>
</thead>
<tbody>
<tr>
<td>The physical location setting in the phone device pool matches the physical location setting in a device pool that is associated with the matching subnet.</td>
<td>The system does not consider the phone to be roaming, and the system uses the settings in the home location device pool.</td>
</tr>
<tr>
<td>Note: Although the phone may have moved from one subnet to another, the physical location and associated services have not changed.</td>
<td></td>
</tr>
<tr>
<td>The matching subnet has a single device pool that is assigned to it; the subnet device pool differs from the home location device pool, and the physical locations differ.</td>
<td>The system considers the phone to be roaming. It reregisters with the parameters of the device pool for the matching subnet.</td>
</tr>
<tr>
<td>The physical locations differ, and the matching subnet has multiple device pools assigned to it.</td>
<td>The system considers the phone to be roaming. The new device pool gets assigned according to a round-robin rule. Each time that a roaming device comes in to be registered for the subnet, the next device pool in the set of available device pools gets assigned.</td>
</tr>
<tr>
<td>Physical location gets defined for the home device pool but is not defined for the device pools that are associated with the matching subnet.</td>
<td>The physical location has not changed, so the phone remains registered in the home device pool.</td>
</tr>
<tr>
<td>Physical location that is not defined for the home device pool gets defined for the device pools that are associated with the matching subnet.</td>
<td>The system considers the phone to be roaming to the defined physical location, and it registers with the parameters of the device pool for the matching subnet.</td>
</tr>
<tr>
<td>A subnet gets updated or removed.</td>
<td>The rules for roaming and assigning device pools get applied by using the remaining subnets.</td>
</tr>
</tbody>
</table>

Note: If no device mobility information entries match the device IP address, the device uses the home location device pool settings.

Device Mobility Groups Operations Summary

You can use device mobility groups to determine when a device moves to another location within a geographic entity, so a user can use its own dial plan. For example, you can configure a device mobility group for the United States and another group for the United Kingdom. If a phone moves into a different mobility group (such as from the United States to the United Kingdom), Unified Communications Manager uses the Calling Search Space, AAR Group and AAR CSS from the phone record, and not from the roaming location.
If the device moves to another location with same mobility group (for example, Richardson, USA, to Boulder, USA), the CSS information gets taken from the roaming device pool settings. With this approach, if the user is dialing PSTN destinations, the user reaches the local gateway.

The following table describes the device pool parameters that the system uses for various scenarios.

**Table 5: Device Mobility Group Scenarios**

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Parameters Used</th>
</tr>
</thead>
</table>
| A roaming device moves to another location in the same device mobility group. | Roaming Device Pool: yes  
Location: Roaming device pool setting  
Region: Roaming device pool setting  
Media Resources Group List: Roaming device pool setting  
Device CSS: Roaming device pool setting (Device Mobility CSS)  
AAR Group: Roaming device pool setting  
AAR CSS: Roaming device pool setting |
| A roaming device moves to another location in a different device mobility group. | Roaming Device Pool: yes  
Location: Roaming device pool setting  
Region: Roaming device pool setting  
Media Resources Group List: Roaming device pool setting  
Device CSS: Home location settings  
AAR Group: Home location settings  
AAR CSS: Home location settings |
| A device roams, and a device mobility group does not get defined for the home or roaming device pool. | Because the device is roaming, it takes the roaming device pool settings, including the Device Mobility Calling Search Space, AAR Calling Search Space, and AAR Group. |

**Device Mobility Prerequisites**

- The phone must have a dynamic IP address to use device mobility. If a phone with a static IP address roams, Unified Communications Manager uses the configuration settings from its home location.

- The Device Mobility feature requires you to set up device pools with site-specific settings. This chapter describes only the device pool settings that relate to device mobility. For more detailed information on configuring device pools, see the "Configure Device Pools" chapter in the System Configuration Guide for Cisco Unified Communications Manager.

- Cisco Database Layer Monitor service must be running on the same node as the Cisco CallManager service.

- Cisco TFTP service must be running on at least one node in the cluster.
- Cisco Unified Communications Manager Locale Installer (if you want to use non-English phone locales or country-specific tones).
- Any phone that runs either SCCP or SIP.

Device Mobility Configuration Task Flow

Complete these tasks to configure device mobility.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | Enable device mobility at the device level by completing either of these tasks:  
- Enable Device Mobility Clusterwide, on page 48  
- Enable Device Mobility for Individual Devices, on page 49 | Enables device support through a clusterwide service parameter, or within the Phone Configuration window of an individual device. |
| **Step 2** | Configure a Physical Location, on page 49 | Set up the physical locations that you will assign to your device pools. |
| **Step 3** | Configure a Device Mobility Group, on page 50 | A device mobility group is a logical grouping of sites with similar dialing patterns. |
| **Step 4** | Configure a Device Pool for Device Mobility, on page 50 | Assign the physical location, device mobility group, and other device mobility-related information to device pools that will be used for device mobility. |
| **Step 5** | Configure Device Mobility Information, on page 51 | Assign the IP subnets where roaming devices can register and the device pools that can be assigned to those roaming devices. |

Enable Device Mobility Clusterwide

Configure a service parameter that sets the default device mobility setting to **On** for all phones clusterwide except where there is an overriding configuration in that phone's Phone Configuration.

**Procedure**

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

**Step 2**  
From the **Server** drop-down list, choose the node that is running the Cisco CallManager service.

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

**Step 4**  
Under **Clusterwide Parameters (Device - Phone)**, set the **Device Mobility Mode** service parameter to **On**.

**Step 5**  
Click **Save**.
For devices that are already registered, you must restart the Cisco CallManager service for this new setting to be enabled.

**What to do next**

If you want to configure device mobility settings for an individual device, go to Enable Device Mobility for Individual Devices, on page 49.

Otherwise, you can begin configuring the system for device mobility. Go to Configure a Physical Location, on page 49.

### Enable Device Mobility for Individual Devices

Use this procedure to enable device mobility for an individual device. This configuration overrides the setting of the Device Mobility Mode clusterwide service parameter.

**Procedure**

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

**Step 2**
Click **Find** and select the device that you want to configure.

**Step 3**
From the **Device Mobility Mode** drop-down list, choose one of the following:

- **On** — Device mobility is enabled for this device.
- **Off** — Device mobility is disabled for this device.
- **Default** — The device uses the setting of the **Device Mobility Mode** clusterwide service parameter. This is the default setting.

**Step 4**
Click **Save**.

### Configure a Physical Location

Use this procedure to configure a physical location that you will assign to a device pool. Device Mobility uses the location of the device registration to assign an appropriate device pool.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **System > Physical Location**.

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

**Step 3**
Enter a **Name** for the location.

**Step 4**
Enter a **Description** for the location.

**Step 5**
Click **Save**.
Configure a Device Mobility Group

A device mobility group is a logical grouping of sites with similar dialing patterns. For example, a company with a worldwide network may want to set up device mobility groups that represent individual countries.

**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; Device Mobility &gt; Device Mobility Group</strong>.</td>
</tr>
<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 device mobility group.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Enter a <strong>Description</strong> for the device mobility group.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

Configure a Device Pool for Device Mobility

Use this procedure to set up a device pool with parameters that you configured for device mobility.

**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; Device Pool</strong>.</td>
</tr>
</tbody>
</table>
| Step 2 | Do either of the following:  
  • Click **Find** and select an existing device pool.  
  • Click **Add New** to create a new device pool. |
| Step 3 | Under **Roaming Sensitive Settings**, assign the parameters that you set up in the previous device mobility tasks:  
  • **Physical Location**—From the drop-down, select the physical location that you set up for this device pool. Device mobility uses this location to assign a device pool for a roaming device.  
  • **Device Mobility Group**—From the drop-down, select the device mobility group that you set up for this device pool. |
| Step 4 | Under **Device Mobility Related Information**, configure the following device mobility-related fields. For more information on the fields and their configuration options, see the system Online Help:  
  • **Device Mobility Calling Search Space**—Select the CSS to be used by a roaming device that uses this device pool.  
  • **AAR Calling Search Space**—Select the calling search space for the device to use when automated alternate routing (AAR) is performed.  
  • **AAR Group**—If AAR is configured, select the AAR Group for this device.  
  • **Calling Party Transformation CSS**—Select the calling party transformation CSS for roaming devices that use this device pool. |
Note

- The **Calling Party Transformation CSS** overrides the device level configuration for roaming devices, even if the **Use Device Pool Calling Party Transformation CSS** check box is unchecked in the **Phone Configuration** window.

- The **Called Party Transformation CSS** setting is applied to the gateway rather than to the roaming device.

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

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

---

**Configure Device Mobility Information**

Use this procedure to configure Device Mobility Info, representing the IP subnets to which roaming devices can register and the corresponding device pools that the system can assign to roaming devices.

**Procedure**

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

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

**Step 3** Enter a **Name** for the Device Mobility Info.

**Step 4** Enter the IP subnet details for roaming device registrations.

- If you are using IPv4 addresses for your mobile devices, complete the IPv4 subnet details.
- If you are using IPv6 addresses for your mobile devices, complete the IPv6 subnet details.

**Step 5** Select the device pools that you want the system to assign for roaming devices that register to one of these subnets. Use the arrows to move the appropriate device pools from the **Selected Device Pools** list box to the **Available Device Pools** list box.

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

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

---

**View Roaming Device Pool Parameters**

Follow this procedure if you want to view and verify the current device mobility settings for a device.

**Procedure**

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

**Step 2** Enter search criteria and click **Find** to find the device with device mobility mode enabled.

**Step 3** Click **View Current Device Mobility Settings** next to the **Device Mobility Mode** field.
The roaming device pool settings appear. If the device is not roaming, the home location settings appear.

**Device Mobility Interactions**

*Table 6: Device Mobility Interactions*

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party</td>
<td>Calling party normalization enhances the dialing capabilities of some phones and improves call-back functionality when a call is routed to multiple geographical locations; that is, the feature ensures that the called party can return a call without the need to modify the directory number in the call log directories on the phone. Additionally, calling party normalization allows you to globalize and localize phone numbers, so the appropriate calling number presentation is displayed on the phone.</td>
</tr>
<tr>
<td>Normalization</td>
<td></td>
</tr>
</tbody>
</table>
| Roaming              | When a device is roaming in the same device mobility group, Unified Communications Manager uses the device mobility CSS to reach the local gateway. If a user sets call forward all (CFA) at the phone, the CFA CSS is set to None, and the CFA CSS activation policy is set to With Activating Device/Line CSS, then the following behaviors will occur, depending on the device location:  
  • The Device CSS and Line CSS are used as the CFA CSS when the device is in its home location.  
  • If the device is roaming within the same device mobility group, the device mobility CSS from the roaming device pool and the line CSS are used as the CFA CSS.  
  • If the device is roaming within a different device mobility group, the Device CSS and Line CSS are used as the CFA CSS. |
### Device Mobility Restrictions

**Table 7: Device Mobility Restrictions**

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| IP Address  | The device mobility feature depends on the IPv4 address or IPv6 address of the device that registers with Unified Communications Manager.  
  - The phone must have a dynamic IPv4 address or IPv6 address to use the device mobility.  
  - If the device is assigned an IP address by using network address translation (NAT) or port address translation (PAT), the IP address that is provided during registration may not match the actual IP address of the device.  
  - If the Cisco IP phone supports IPv4-Only Stack or IPv6-Only Stack, then the phone gets re-associated either with IPv4 or IPv6 Device Mobility Information, based on the IP addressing mode preference defined. For example, when a phone is defined with IPv6 preference but has no matching Device Mobility Information (IPv6 subnet and mask size), then it is associated with IPv4. When you add matching IPv6 Device Mobility Information, then the phone gets re-associated with IPv6 Device Mobility Information. |
Extend and Connect Overview

The Extend and Connect feature allows administrators to deploy Unified Communications Manager (UC) Computer Telephony Integration (CTI) applications that interoperate with any endpoint. With Extend and Connect, users can access UC applications from any location using any device.

The Extend and Connect feature for Unified Communications Manager provides the following UC features:

- Receive incoming enterprise calls
- Make Call
- Disconnect
- Hold and Retrieve
- Redirect and Forward
- Call Forward All
- Call Forward Busy
- Call Forward No Answer
- Do Not Disturb
- Play Dual Tone Multi Frequency (DTMF) (out-of-band and in-band)
- Consult Transfer, Conference
- Add, edit, and delete remote destinations
- Set remote destination as Active or Inactive
### Extend and Connect Prerequisites

- Cisco Jabber, Release 9.1(1) or later
- Cisco Unified Workspace License (CUWL) Standard, CUWL Professional, or Cisco User Connect License (UCL) - Enhanced

### Extend and Connect Configuration Task Flow


**Before you begin**
- Review *Extend and Connect Prerequisites*, on page 56.

**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 User Account, on page 57</strong></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>Add User Permissions, on page 57</strong></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>Create CTI Remote Devices, on page 58</strong></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td><strong>Add Directory Number to a Device, on page 59</strong></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>Add Remote Destination, on page 59</strong></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>Verify Remote Destination, on page 60</strong></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td><strong>Associate User with Device, on page 61</strong></td>
</tr>
</tbody>
</table>
Configure User Account

For a new or existing user in Unified Communications Manager, you must enable user mobility so that they can use CTI remote devices. If you do not enable mobility for users, you cannot assign those users as owners of CTI remote devices.

Procedure

Step 1  From Cisco Unified CM Administration, choose User Management > End User. The Find and List Users window appears.

Step 2  Perform one of the following:
   • Click Add New, to configure a new user.
   • Apply the filters using the Find User Where field and click Find to retrieve a list of users.
   Note You may add the new end user account through LDAP integration or local configuration.

The End User Configuration window appears.

Step 3  Locate the Mobility Information section.
Step 4  Check the Enable Mobility check box.
Step 5  Click Save.

Add User Permissions

After the end user is active in Unified Communications Manager, add access control group permissions.

Procedure

Step 1  From Cisco Unified CM Administration, choose User Management > End User.
Step 2  Specify the appropriate filters in the Find User Where field, and then select Find to retrieve a list of users.
Step 3  Select the user from the list.

The End User Configuration window appears.

Step 4  Locate the Permissions Information section.
Step 5  Click Add to Access Control Group.

The Find and List Access Control Groups window appears.

Step 6  Click Find.

The Access Control Group list for Standard Users appears.

Step 7  Check the check boxes next to the following permissions:
   • Standard CCM End-Users
Create CTI Remote Devices

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.

Unified Communications Manager provides Extend and Connect capabilities to control calls on devices such as public switched telephone network (PSTN) phones and private branch exchange (PBX) devices.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Add New.
Step 3 Select CTI Remote Device from the Phone Type drop-down list and then click Next.

The Phone Configuration window appears.

Step 4 Select the appropriate user ID from the Owner User ID drop-down list.

Note Only users for whom you enable mobility are available from the Owner User ID drop-down list.

Unified Communications Manager populates the Device Name field with the user ID and a CTRID prefix, for example, CTRIDusername.

Step 5 Edit the default value in the Device Name field, if appropriate.
Step 6 Enter a meaningful description in the Description field.

Note Cisco Jabber displays device descriptions to users. If Cisco Jabber users have multiple devices of the same model, the descriptions from Unified Communications Manager help users tell the difference between them.

Step 7 Ensure that you select an appropriate option from the Rerouting Calling Search Space drop-down list in the Protocol Specific Information section.

The Rerouting Calling Search Space drop-down list defines the calling search space for rerouting and ensures that users can send and receive calls from the CTI remote device.

Step 8 Configure the remaining fields in the Phone Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 9 Click Save.

The fields to associate directory numbers and add remote destinations are displayed in the Phone Configuration window.
Add Directory Number to a Device

A directory number (DN) is a numerical address that is configured as a line on the CTI remote device. A DN typically represents the primary work number of a user (for example, 2000 or +1 408 200 2000).

Note
The Calling Search Space (CSS) and partition of DN are mandatory on devices.

Note
The CTI Remote Device should not block its own DN. The CSS is important for the CTIRD device to reach its own DN.

Follow these steps to add a directory number to a CTI remote device.

Procedure

Step 1 Locate the Association Information section in the Phone Configuration window.
Step 2 Click Add a new DN.
The Directory Number Configuration window appears.
Step 3 Specify a directory number in the Directory Number field.
Step 4 Configure all other required fields. For more information on the fields and their configuration options, see the system Online Help.
Step 5 Click Save.

Add Remote Destination

A remote destination is a numerical address or directory URI that represents the other phones that the user owns (for example, a home office line or other PBX phone). A remote destination may be any off-cluster device. Unified Communications Manager automatically applies application dial rules to all remote destination numbers for CTI remote devices. By default, four remote destinations are supported per device. You can set the maximum number to 10 remote destinations per device in End User Configuration window.

Note
You can determine which remote destination the Jabber client has set as Active by opening the Phone Configuration window from the Cisco Unified Communications Manager Administration interface.

- Unified Communications Manager automatically verifies whether it can route calls to remote destinations that Cisco Jabber users add through the client interface.
- Unified Communications Manager does not verify whether it can route calls to remote destinations that you add through the Cisco Unified Communications Manager Administration interface.

**Procedure**

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

**Step 2** Specify the appropriate filters in the **Find Phone Where** field to and then click **Find** to retrieve a list of phones.

**Step 3** Select the CTI remote device from the list.

The **Phone Configuration** window appears.

**Step 4** Locate the **Associated Remote Destinations** section.

**Step 5** Click **Add a New Remote Destination**.

The **Remote Destination Information** window appears.

**Step 6** Enter the destination number in the **Destination Number** field.

To use the remote destination with Cisco Jabber clients, you must configure the destination name as **JabberRD**.

**Step 7** Configure the remaining fields in the **Remote Destination Information** window.

**Step 8** Click For more information on the fields and their configuration options, see the system Online Help.

**Verify Remote Destination**

Perform these steps to verify if the remote destination is successfully added for a user.

**Procedure**

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

**Step 2** Specify the appropriate filters in the **Find Phone Where** field to and then click **Find** to retrieve a list of phones.

**Step 3** Select the CTI remote device from the list.

The **Phone Configuration** window appears.

**Step 4** Locate the **Associated Remote Destinations** section and verify that the remote destination is available.
Step 5  Click Apply Config.

Note  The Device Information section on the Phone Configuration window indicates when a remote destination is active or controlled by Cisco Jabber.

Associate User with Device

Procedure

Step 1  From Cisco Unified CM Administration, choose User Management > End User.
Step 2  Specify the appropriate filters in the Find User Where field to and then click Find to retrieve a list of users.
Step 3  Select the user from the list. The End User Configuration window appears.
Step 4  Locate the Device Information section.
Step 5  Click Device Association. The User Device Association window appears.
Step 6  Find and select the CTI remote device.
Step 7  To complete the association, click Save Selected/Changes.
Step 8  From Related Links drop-down list, choose Back to User, and then click Go. The End User Configuration window appears, and the associated device that you chose appears in the Controlled Devices pane.

CTI Remote Device (CTIRD) Call Flows

Unified Communications Manager supports the separate calling party number and billing number feature when users are added as CTI Remote Device. Each CTI Remote Device is configured with the user directory number (DN) (for example, 2000) and a remote destination that represents any off-cluster device (for example, a PBX phone with the number +1 408 111 1111).

When a call is initiated from the PSTN network to a CTIRD line, Unified Communications Manager looks for FROM and PAID header information. The FROM header contains the external presentation name and number and PAID contains the identity of the user (that is a user’s DN or DDI).

If FROM and PAID headers have different numbers and Enable External Presentation Name and Number check box is checked in its SIP profile and Display External Presentation Name and Number service parameter value set to True, then Unified Communications Manager displays the FROM header information on the called device. In the same way, if any one option is disabled, Unified Communications Manager displays PAID header information on the called device.

Similarly, in the outgoing call scenario a user calls from Remote Destination (CTIRD line) configured with External Presentation Name and External Presentation Number on Directory Number configuration page to a PSTN network through a SIP trunk with Enable External Presentation Name and Number configured in its SIP profile. Then, Unified Communications Manager send the External Presentation Information.
configured on the Directory Number Configuration page in the FROM header of the outgoing SIP message to display on the called device.

If Enable External Presentation Name and Number check box is unchecked, then Unified Communications Manager sends the directory number information in the FROM and PAID to display on the called device and configured External Presentation Information in the X-Cisco-Presentation header.

If you check the Anonymous External Presentation check box, the configured External Presentation Name and Number are removed from the respective fields and external presentation displayed as anonymous on the called device.

For more details on Configuring External Presentation Information, see the Configure Directory Number chapter in the System Configuration Guide for Cisco Unified Communications Manager.

### Extend and Connect Interactions

**Table 8: Extend and Connect Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory URI Dialing</td>
<td>Configure a Directory URI as the DN, remote destination, or both for the CTI remote device.</td>
</tr>
<tr>
<td>Unified Mobility</td>
<td>Extend and Support does not support moving active calls between a Cisco Unified IP Phone and a remote destination. If you want the capabilities of both Unified Mobility and Extend and Connect, you can configure the same remote destination on the Remote Device Profile and CTI Remote Device types when the Owner ID of both device types is the same. This configuration allows Cisco Mobility features to be used concurrently with Extend and Connect. The ability to configure the same remote destination on both device types is supported using Cisco Unified Communications Manager Release 10.0(1) or later. Do not configure remote destinations that are used with the Cisco Extend and Connect feature on Cisco Dual-mode for iPhone, Cisco Dual-mode for Android, and Carrier-integrated Mobile device types. Do not use prefixes to differentiate the same remote destination address. For example, 91-4085555555 and +1-4085555555 are treated as the same number.</td>
</tr>
</tbody>
</table>
| Hunt List               | The Extend and Connect feature allows users to receive hunt calls on remote destination phones under the following conditions:  
  • The user has a Cisco Unified IP Phone.  
  • The Cisco Unified P Phone is available to answer hunt calls (logged-in/HLog).  
  • Cisco Jabber is running in Extend and Connect mode. |
### Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1

#### Extend and Connect Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum number of remote destinations</td>
<td>You can configure up to ten remote destinations for each CTI remote device.</td>
</tr>
<tr>
<td></td>
<td>By default, four remote destinations are supported per device. You can set</td>
</tr>
<tr>
<td></td>
<td>the maximum number to 10 remote destinations per device.</td>
</tr>
<tr>
<td>Off-cluster devices</td>
<td>• Remote destination numbers must represent off-cluster devices.</td>
</tr>
<tr>
<td></td>
<td>• Remote destinations can be off-cluster URIs.</td>
</tr>
<tr>
<td>Directory numbers</td>
<td>You cannot configure directory numbers as remote destination numbers.</td>
</tr>
<tr>
<td>Cisco Jabber</td>
<td>Before you save the remote destinations that are configured using Cisco</td>
</tr>
<tr>
<td></td>
<td>Jabber, verify if the remote destinations can be routed by the configured</td>
</tr>
<tr>
<td></td>
<td>dial plan.</td>
</tr>
<tr>
<td>Application dial rules</td>
<td>Application Dial Rules are applied to all remote destinations that are</td>
</tr>
<tr>
<td></td>
<td>configured on the CTI remote device through the Cisco Unified Communications</td>
</tr>
<tr>
<td></td>
<td>Manager Administration interface and Cisco Jabber.</td>
</tr>
<tr>
<td></td>
<td>Advise end users which number formats the Application Dial Rules are</td>
</tr>
<tr>
<td></td>
<td>configured to support (for example, nn-nnn-nnnn, E.164, both).</td>
</tr>
<tr>
<td>Remote destination number</td>
<td>Each remote destination number must be unique within the cluster.</td>
</tr>
<tr>
<td></td>
<td>The same remote destination number cannot be used by two or more users.</td>
</tr>
</tbody>
</table>
Remote Worker Features

### Remote Worker Features

**Remote Worker Features**

**Description**

- Remote destination numbers are validated using the CTI remote device reroute calling search space.
- Remote destinations that are configured using the Cisco Unified Communications Manager Administration interface and AXL interface are not validated.

**Call Forward Unregistered**

Extend and Connect does not support Call Forward Unregistered Internal or Call Forward Unregistered External.

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Remote destination validation</td>
<td>• Remote destination numbers are validated using the CTI remote device reroute calling search space.</td>
</tr>
<tr>
<td></td>
<td>• Remote destinations that are configured using the Cisco Unified Communications Manager Administration interface and AXL interface are not validated.</td>
</tr>
<tr>
<td>Call Forward Unregistered</td>
<td>Extend and Connect does not support Call Forward Unregistered Internal or Call Forward Unregistered External.</td>
</tr>
</tbody>
</table>
Remote Worker Emergency Calling Overview

The Remote Worker Emergency Calling feature enables customers to provide reliable emergency calling support to remote workers by using remote Virtual Private Network (VPN) connections. Emergency calls from off-premises users are routed to the Public Safety Answering Point (PSAP), and user-provided location information is delivered with each call.

To use this feature, remote workers must confirm or update their location whenever their device registration is interrupted. A customizable disclaimer notice is first displayed on the devices that are designated for off-premises (connected remotely to the customer network), which advises the users to provide correct location information. After the location information is provided, the off-premises location that is currently associated with the designated device is displayed. Users can confirm their current location or select another previously stored location from their device display; if their location is new, they are directed to the Cisco Emergency Responder Off-Premises User web page to create a new location.

Before completing this process, the administrator may restrict the device to calling a single configured destination. This action ensures that the device user has acknowledged the disclaimer and provided current location information before the device is enabled for normal use.

Remote Worker Emergency Calling Prerequisites

You must configure Intrado (a third party application) on the Cisco Emergency Responder before you configure the Remote Worker Emergency Calling feature. For information about configuring Intrado on the Cisco Emergency Responder, see Cisco Emergency Responder Administration Guide.
Before you begin

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure User As a Remote Worker, on page 66</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Specify Alternate Routing for Emergency Calling, on page 67</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure the Application Server, on page 67</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure E911 Messages, on page 67</td>
</tr>
</tbody>
</table>

Configure User As a Remote Worker

Before you begin

Ensure that you have configured Intrado on the Cisco Emergency Responder. For more information about configuring Intrado on the Cisco Emergency Responder, see.

Procedure

- **Step 1** From Cisco Unified CM Administration, choose Device > Phone.
- **Step 2** Enter the appropriate search criteria to find the phone and click Find. A list of phones that match the search criteria is displayed.
- **Step 3** Choose the phone for which you want to configure Remote Worker Emergency Calling. The Phone Configuration window is displayed.
- **Step 4** From the Device Information section, select the appropriate user ID from the Owner User ID drop-down list and check the Remote Device check box.
- **Step 5** Click Save.
Specify Alternate Routing for Emergency Calling

Perform the following steps to configure calling search space and destination number. These parameters are used to restrict the routing of any call made from a registered off-premises device where the user has not set a location. If you do not configure these parameters, the calls are routed normally.

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; Service Parameters.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list, choose a server.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Service drop-down list, choose Cisco CallManager. The Service Parameter Configuration window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Clusterwide Parameters (Emergency Calling for Required Off-premise Location) section, specify Alternate Destination for Emergency Call.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Specify Alternate Calling Search Space for Emergency Call.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Configure the Application Server

You must configure the application server to enable the E911 Proxy to communicate with the Cisco Emergency Responder. E911 proxy is used to direct the users to the application server where they enter the location of the device.

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; Application Server.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New. The Application Server window appears.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Application Server Type drop-down list, select CER Location Management.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Next.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the Name field, specify a name to identify the application server that you are configuring.</td>
</tr>
<tr>
<td>Step 6</td>
<td>In the IP address field, specify the IP address of the server that you are configuring.</td>
</tr>
<tr>
<td>Step 7</td>
<td>From the list of Available Application Users, select the application user and click the Down arrow.</td>
</tr>
<tr>
<td>Step 8</td>
<td>In the End User URL field, enter a URL for the end users that are associated with this application server.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Configure E911 Messages

Use the following procedure to select and edit E911 messages for off-premises devices.
Procedure

**Step 1**  From Cisco Unified CM Administration, choose **System > E911 Messages**.

**Step 2**  Select the required language link of the E911 messages. The **E911 Messages Configuration** page displays the Agreement, Disclaimer, and Error messages.

**Step 3**  (Optional) Edit the E911 messages to be displayed on off-premises devices.

**Step 4**  Click **Save**.
Configure Mobile and Remote Access

• Mobile and Remote Access Overview, on page 69
• Mobile and Remote Access Prerequisites, on page 71
• Mobile and Remote Access Configuration Task Flow, on page 72

Mobile and Remote Access Overview

Unified Communications Manager Mobile and Remote Access is a core part of the Cisco Collaboration Edge Architecture. It allows endpoints such as Cisco Jabber to have their registration, call control, provisioning, messaging, and presence services that are provided by Unified Communications Manager when the endpoint is not within the enterprise network. Cisco Expressway connects the mobile endpoint to the on-premises network, providing secure firewall traversal and line-side support for Unified CM registrations.

The overall solution provides:

• Off-premises access: a consistent experience outside the network for Jabber and EX/MX/SX Series clients
• Security: secure business-to-business communications
• Cloud services: enterprise grade flexibility and scalable solutions providing rich Cisco Webex integration and Service Provider offerings
• Gateway and interoperability services: media and signaling normalization, and support for non-standard endpoints
Third-party SIP or H.323 devices can register to the Expressway-C and, if necessary, interoperate with Unified CM-registered devices over a SIP trunk.

**Figure 3: Typical Call Flow: Signaling and Media Paths**

- Unified CM provides call control for both mobile and on-premises endpoints.
- Signaling traverses the Expressway solution between the mobile endpoint and Unified CM.
- Media traverses the Expressway solution and is relayed between endpoints directly; all media is encrypted between the Expressway-C and the mobile endpoint.

**Configuring Mobile and Remote Access**

To enable Cisco Jabber users with MRA functionality, set up an MRA User Policy within the User Profile Configuration window of Unified Communications Manager. The MRA User Policy is not required for non-Jabber endpoints.

In addition, you must configure Cisco Expressway with Mobile and Remote Access. For details, see Mobile and Remote Access via Cisco Expressway Deployment Guide.
Mobile and Remote Access Prerequisites

Cisco Unified Communications Manager Requirements

The following requirements apply:

- If you are deploying multiple Unified Communications Manager clusters, set up an ILS network.
- Mobile and Remote Access requires that you set up NTP servers for your deployment. Make sure that you have NTP servers deployed for your network and Phone NTP References for SIP endpoints.
- If you are deploying ICE for media path optimization, you will need to deploy a server that can provide TURN and STUN services.

DNS Requirements

For the internal connection to Cisco Expressway, configure the following locally resolvable DNS SRV that points to Unified Communications Manager:

```
_cisco-uds._tcp<domain>
```

You must create internal DNS records, for both forward and reverse lookups, for all Unified Communications nodes used with Mobile and Remote Access. This allows Expressway-C to find the nodes when IP addresses or hostnames are used instead of FQDNs. Make sure that the SRV record is not resolvable outside of the local network.

Cisco Expressway Requirements

This feature requires you to integrate Unified Communications Manager with Cisco Expressway. For Cisco Expressway configuration details for Mobile and Remote Access, refer to the Mobile and Remote Access Through Cisco Expressway Deployment Guide.

The minimum Expressway release for MRA Access Policy support with Cisco Jabber is X8.10.

Certificate Prerequisites

You must exchange certificates between Unified Communications Manager, the IM and Presence Service, and Cisco Expressway-C. Cisco recommends that you use CA-signed certificates with the same CA for each system. In this case:

- Install the CA root certificate chain on each system (for Unified Communications Manager and the Instant Messaging and Presence Service install the certificate chain to the tomcat-trust store).
- For Unified Communications Manager, issue a CSR to request CA-signed tomcat (for AXL and UDS traffic) and Cisco CallManager (for SIP) certificates.
- For the Instant Messaging and Presence Service, issue a CSR to request CA-signed tomcat certificates.

If you use different CAs, you must install each CA's root certificate chain on Unified Communications Manager, Instant Messaging and Presence Service, and Expressway-C.
You can also use self-signed certificates for both Unified Communications Manager and the Instant Messaging and Presence Service. In this case, you must upload onto Expressway-C the tomcat and Cisco CallManager certificates for Unified Communications Manager and a tomcat certificate for the Instant Messaging and Presence Service.

## Mobile and Remote Access Configuration Task Flow

Complete these tasks in Unified Communications Manager if you want to deploy Mobile and Remote Access endpoints.

### 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 Cisco AXL Web Service, on page 73</td>
<td>Make sure that the Cisco AXL Web Service is activated on the publisher node.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Maximum Session BitRate for Video, on page 73</td>
<td><strong>Optional.</strong> Configure Region-specific settings for your MRA endpoints. For example, if you expect MRA endpoints to use video, you may want to increase the Maximum Session Bit Rate for Video Calls setting as the default setting of 384 kbps may be too low for some video endpoints.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Device Pool for MRA, on page 73</td>
<td>Assign your Date/Time Group and Region configuration to the device pool that your MRA endpoints use.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure ICE, on page 74</td>
<td><strong>Optional.</strong> ICE is an optional deployment that uses STUN and TURN services to analyze the available media paths for an MRA call and then to select the best path. ICE adds potentially to the call setup time, but increases the reliability of MRA calls.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure Phone Security Profile for MRA, on page 75</td>
<td>Use this procedure to set up a phone security profile to be used by MRA endpoints.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Configure MRA Access Policy for Cisco Jabber Users, on page 76</td>
<td>Cisco Jabber only. Set up an MRA Access Policy for Cisco Jabber users. Cisco Jabber users must be enabled with MRA access within their user profiles in order to use the MRA feature.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Configure Users for MRA, on page 77</td>
<td>For Cisco Jabber users, the User Policy that you set up must be applied to their End User Configurations.</td>
</tr>
</tbody>
</table>
### Activate Cisco AXL Web Service

Make sure that the Cisco AXL Web Service is activated on the publisher node.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>From Cisco Unified Serviceability, choose <strong>Tools &gt; Service Activation</strong>.</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>From the <strong>Server</strong> drop-down list, select the publisher node and click <strong>Go</strong>.</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Under <strong>Database and Admin Services</strong>, confirm that the <strong>Cisco AXL Web Service</strong> is <strong>Activated</strong>.</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>If the service is not activated, check the corresponding check box and click <strong>Save</strong> to activate the service.</td>
<td></td>
</tr>
</tbody>
</table>

### Configure Maximum Session BitRate for Video

Configure Region settings for your MRA endpoints. The default settings may be sufficient in many cases, but if you expect MRA endpoints to use video, you may want to increase the **Maximum Session Bit Rate for Video Calls** within your Region Configuration. The default setting of 384 kbps may be too low for some video endpoints, such as the DX series.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>From Cisco Unified CM Administration, choose <strong>System &gt; Region Information &gt; Region</strong>.</td>
<td></td>
</tr>
</tbody>
</table>
| 2    | Perform any one of the following:  
  - Click **Find** and select the region to edit the bit rates within an existing region.  
  - Click **Add New** to create a new region. |
| 3    | In the **Modify Relationship to other Regions** area, configure a new setting for the **Maximum Session Bit Rate for Video Calls**. For example, 6000 kbps. |
| 4    | Configure any other fields in the **Region Configuration** window. For more information on the fields and their configuration options, see the system **Online Help**. |
| 5    | Click **Save**. |

### Configure a Device Pool for MRA

When you created a new region, assign your region to the device pool that your MRA endpoints use.
Configure ICE

Use this procedure if you want to deploy ICE to handle call setup for MRA calls. ICE is an optional deployment that uses STUN and TURN services to analyze the available media paths for an MRA call and to select the best path. ICE adds potentially to the call setup time, but increases the reliability of MRA calls.

Before you begin

Decide how you are going to deploy ICE. You can configure ICE for groups of phones via the Common Phone Profile Configuration, to individual Cisco Jabber desktop devices, or through system-wide defaults that apply to all phones.

As a fallback mechanism, ICE can use a TURN server to relay media. Make sure that you have deployed a TURN server.

Procedure

Step 1 From Cisco Unified CM Administration:
- Choose System > Enterprise Phone to configure system defaults for ICE.
- Choose Device > Device Settings > Common Phone Profile to configure ICE for groups of endpoints and select the profile you want to edit.
- Choose Device > Phone to configure ICE for an individual Cisco Jabber desktop endpoint and select the endpoint that you want to edit.

Step 2 Scroll down to the Interactive Connectivity Establishment (ICE) section:

Step 3 Set the ICE drop-down list to Enabled.

Step 4 Set the Default Candidate Type:
- Host—A candidate obtained by selecting the IP address on the host device. This is the default.
• **Server Reflexive**—An IP address and port candidate obtained by sending a STUN request. In many cases, this may represent the public IP address of the NAT.
• **Relayed**—An IP address and port candidate obtained from a TURN server. The IP address and port are resident on the TURN server such that media is relayed through the TURN server.

**Step 5** From the **Server Reflexive Address** drop-down list, select whether you want to enable STUN-like services by setting this field to **Enabled** or **Disabled**. You must set this field to enabled if you configured Server Reflexive as the Default Candidate.

**Step 6** Enter the IP address or hostname for the Primary and Secondary TURN Servers.

**Step 7** Set the **TURN Server Transport Type** to **Auto (default setting)**, **UDP**, **TCP**, or **TLS**.

**Step 8** Enter the **Username** and **Password** of the TURN Server.

**Step 9** Click **Save**.

**Note** If you configured ICE for a Common Phone Profile, you must associate phones to that Common Phone Profile for phones to be able to use the profile. You can apply the profile to a phone through the **Phone Configuration** window.

---

**Configure Phone Security Profile for MRA**

Use this procedure to set up a phone security profile to be used by MRA endpoints.

**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, select your device type. For example, you could select **Cisco Unified Client Service Framework** for a Jabber application.

**Step 4** Click **Next**.

**Step 5** Enter a **Name** for the profile. For MRA, the name must be in FQDN format and must include the enterprise domain.

**Note** This field must be set to **Encrypted**. Otherwise, Expressway rejects communications.

**Step 7** Set the **Transport Type** to **TLS**.

**Step 8** Leave the **TFTP Encrypted Config** check box unchecked for the following phones as MRA will not work for these phones with this option enabled: DX Series, IP Phone 7800, or IP Phone 8811, 8841, 8845, 8861 and 8865.

**Step 9** Complete the remaining fields in the **Phone Security Profile Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

**Step 10** Click **Save**.

**Note** You must apply this profile to the Phone Configuration for each of your MRA endpoints.
Configure MRA Access Policy for Cisco Jabber Users

Use this procedure to set up an MRA Access Policy for Cisco Jabber users. Cisco Jabber users must be enabled with MRA access within their user profiles in order to use the MRA feature. The minimum Expressway release for MRA Access Policy support with Cisco Jabber is X8.10.

Note
The MRA Access Policy is not required for non-Jabber users.

For more information on user profiles, see "User Profile Overview" section in System Configuration Guide for Cisco Unified Communications Manager.

Procedure

Step 1
From Cisco Unified CM Administration, choose User Management > User Settings > 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
If you want Cisco Jabber users associated with this user profile, to be able to use the Mobile and Remote Access (MRA) feature, check the Enable Mobile and Remote Access check box.

Note
By default, this check box is selected. When you uncheck this check box, the Jabber Policies section is disabled and No Service client policy option is selected by default.

Note
This setting is mandatory only for Cisco Jabber users. Non-Jabber users do not need this setting to be able to use MRA. MRA feature is applicable only for the Jabber MRA users and not to any other endpoints or clients.

Step 8
Assign the Jabber policies for this user profile. From the Jabber Desktop Client Policy, and Jabber Mobile Client Policy drop-down list, choose one of the following options:
   • No Service—This policy disables access to all Cisco Jabber services.
   • IM & Presence only—This policy enables only instant messaging and presence capabilities.
   • IM & Presence, Voice and Video calls—This policy enables instant messaging, presence, voicemail, and conferencing capabilities for all users with audio or video devices. This is the default option.

Note
Jabber desktop client includes Cisco Jabber for Windows users and Cisco Jabber for Mac users. Jabber mobile client includes Cisco Jabber for iPad and iPhone users and Cisco Jabber for Android users.
Step 9  If you want the users in this user profile to set the maximum login time for Extension Mobility or Extension Mobility Cross Cluster through Cisco Unified Communications Self Care Portal, check the **Allow End User to set their Extension Mobility maximum login time** check box.

**Note**  By default **Allow End User to set their Extension Mobility maximum login time** check box is unchecked.

Step 10  Click **Save**.

---

**Configure Users for MRA**

For Cisco Jabber users, the MRA access policy that you configured must be associated to your Cisco Jabber users during the LDAP sync. For more information on how to provision end users, see "End User Configuration" section in System Configuration Guide for Cisco Unified Communications Manager.

**Configure Endpoints for MRA**

Provision and configure endpoints for Mobile and Remote Access:

- For Cisco Jabber clients, refer to "Cisco Jabber Configuration Task Flow" section in System Configuration Guide for Cisco Unified Communications Manager.

- For other endpoints, refer to "Endpoint Device Configuration" section in System Configuration Guide for Cisco Unified Communications Manager.

**Configure Cisco Expressway for Mobile and Remote Access**

For details on how to configure Cisco Expressway for Mobile and Remote Access, refer to the Mobile and Remote Access Through Cisco Expressway Deployment Guide.
Configure Cisco Expressway for Mobile and Remote Access
PART III

Remote Network Access

- Wireless LAN, on page 81
- WiFi Hotspot, on page 85
- VPN Client, on page 87
Wireless LAN

- Wireless LAN Overview, on page 81
- Wireless LAN Configuration Task Flow, on page 81

Wireless LAN Overview

This feature removes the need for users to configure WiFi parameters on their phones. You can configure WiFi profiles for them. Devices can then automatically download and apply the WiFi configuration from your system. You can configure a network access profile, which contains further security layers that are related to VPN connectivity and HTTP proxy settings.

Wireless LAN 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>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that wireless LAN profiles.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure a Network Access Profile, on page 82</td>
<td>Optional: Configure a network access profile if you want to configure VPN and HTTP proxy settings that you can link to a wireless LAN profile.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Wireless LAN Profile, on page 82</td>
<td>Configure a wireless LAN profile with common WiFi settings to apply to devices or device pools in the enterprise.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure a Wireless LAN Profile Group, on page 82</td>
<td>Group wireless LAN profiles together.</td>
</tr>
<tr>
<td>Step 5</td>
<td>To Link a Wireless LAN Profile Group to a Device or Device Pool, on page 83, perform one of the following subtasks:</td>
<td>After you complete the device link, TFTP adds the wireless LAN profile group to the existing device configuration file, which the device (or</td>
</tr>
</tbody>
</table>
Configure a Network Access Profile

Configure a network access profile if you want to configure VPN and HTTP proxy settings that you can link to a wireless LAN profile.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Network Access Profile
Step 2 Click Add New.
Step 3 Configure the fields in the Network Access Profile Configuration window. For more information on the fields and their configuration options, see the system Online Help.
Step 4 Click Save.

Configure a Wireless LAN Profile

Configure a wireless LAN profile with common WiFi settings to apply to devices or device pools in enterprise.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Wireless LAN Profile
Step 2 Click Add New.
Step 3 Configure the fields in the Wireless LAN Profile Configuration window. For more information on the fields and their configuration options, see the system Online Help.
Step 4 Click Save.

Configure a Wireless LAN Profile Group

Group your wireless LAN profiles.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Wireless LAN Profile Group.
Step 2 Click Add New.
Step 3 Configure the fields in the **Wireless LAN Profile Group Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 4 Click **Save**.

---

**Link a Wireless LAN Profile Group to a Device or Device Pool**

After you complete the device link, TFTP adds the wireless LAN profile group to the existing device configuration file, which the device (or devices tied to a device pool) proceeds to download.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Link a Wireless LAN Profile Group to a Device, on page 83</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Link a Wireless LAN Profile Group to a Device Pool, on page 83</td>
</tr>
</tbody>
</table>

**Link a Wireless LAN Profile Group to a Device**

**Procedure**

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

- **Step 2** Perform one of the following tasks:
  - Enter search criteria, click **Find**, and choose an existing device from the resulting list.
  - Click **Add New**, and choose the device type from the **Phone Type** drop-down list.

- **Step 3** From the **Wireless LAN Profile Group** drop-down list, choose a wireless LAN profile group that you created.

- **Step 4** Click **Save**.

**Link a Wireless LAN Profile Group to a Device Pool**

If you link a wireless LAN profile group at the device and device pool level, your system uses the device pool setting.

**Procedure**

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

- **Step 2** Perform one of the following tasks:
  - Enter search criteria, click **Find**, and choose an existing device pool from the resulting list.
  - Click **Add New**.

- **Step 3** From the **Wireless LAN Profile Group** drop-down list, choose a wireless LAN profile group that you created.
Step 4  Click Save.
WiFi Hotspot

- WiFi Hotspot Overview, on page 85
- Configure Wi-Fi Hotspot Profile, on page 85

WiFi Hotspot Overview

This feature allows users to use their desk phones to provide a WiFi Hotspot, so that they can connect a WiFi device such as a tablet or a smartphone to the network through the desk phone. The desk phones can automatically download the WiFi Hotspot configuration from Unified Communications Manager, and the configuration is then applied to these devices.

Configure Wi-Fi Hotspot Profile

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; Device Settings &gt; Wi-Fi Hotspot Profile.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the fields in the Wi-Fi Hotspot Profile Configuration window. For more information on the fields and their configuration options, see the system Online Help.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>
Configure Wi-Fi Hotspot Profile
CHAPTER 10

VPN Client

- VPN Client Overview, on page 87
- VPN Client Prerequisites, on page 87
- VPN Client Configuration Task Flow, on page 88

VPN Client Overview

The Cisco VPN Client for Cisco Unified IP Phone creates a secure VPN connection for employees who telecommute. All settings of the Cisco VPN Client are configured through Cisco Unified Communications Manager Administration. After the phone is configured within the Enterprise, the users can plug it into their broadband router for instant connectivity.

Note

The VPN menu and its options are not available in the U.S. export unrestricted version of Unified Communications Manager.

VPN Client Prerequisites

Pre-provision the phone and establish the initial connection inside the corporate network to retrieve the phone configuration. You can make subsequent connections using VPN, as the configuration is already retrieved on the phone.
## VPN Client Configuration Task Flow

### Before you begin

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Complete Cisco IOS Prerequisites, on page 89</td>
<td>Complete Cisco IOS prerequisites. Perform this action if you want to configure Cisco IOS VPN.</td>
</tr>
<tr>
<td>2</td>
<td>Configure Cisco IOS SSL VPN to Support IP Phones, on page 89</td>
<td>Configure Cisco IOS for VPN client on an IP Phone. Perform this action if you want to configure Cisco IOS VPN.</td>
</tr>
<tr>
<td>3</td>
<td>Complete ASA Prerequisites for AnyConnect, on page 91</td>
<td>Complete ASA prerequisites for AnyConnect. Perform this action if you want to configure ASA VPN.</td>
</tr>
<tr>
<td>4</td>
<td>Configure ASA for VPN Client on IP Phone, on page 91</td>
<td>Configure ASA for VPN client on an IP Phone. Perform this action if you want to configure ASA VPN.</td>
</tr>
<tr>
<td>5</td>
<td>Configure the VPN concentrators for each VPN Gateway.</td>
<td>To avoid long delays when the user upgrades the firmware or configuration information on a remote phone, set up the VPN concentrator close in the network to the TFTP or Unified Communications Manager server. If this is not feasible in your network, you can set up an alternate TFTP or load server that is next to the VPN concentrator.</td>
</tr>
<tr>
<td>6</td>
<td>Upload VPN Concentrator Certificates, on page 94</td>
<td>Upload the VPN concentrator certificates.</td>
</tr>
<tr>
<td>7</td>
<td>Configure VPN Gateway, on page 94</td>
<td>Configure the VPN gateways.</td>
</tr>
<tr>
<td>8</td>
<td>Configure VPN Group, on page 95</td>
<td>After you create a VPN group, you can add one of the VPN gateways that you just configured to it.</td>
</tr>
<tr>
<td>9</td>
<td>Perform one of the following:</td>
<td>You must configure a VPN profile only if you have multiple VPN groups. The VPN Profile fields take precedence over the VPN Feature Configuration fields.</td>
</tr>
<tr>
<td></td>
<td>• Configure VPN Profile, on page 96</td>
<td>• Configure VPN Feature Parameters, on page 97</td>
</tr>
<tr>
<td>10</td>
<td>Add VPN Details to Common Phone Profile, on page 99</td>
<td>Add the VPN Group and VPN Profile to a Common Phone Profile.</td>
</tr>
<tr>
<td>11</td>
<td>Upgrade the firmware for Cisco Unified IP Phone to a version that supports VPN.</td>
<td>To run the Cisco VPN client, a supported Cisco Unified IP Phone must be running firmware release 9.0(2) or higher. For more information</td>
</tr>
</tbody>
</table>
Purpose

about upgrading the firmware, see *Cisco Unified IP Phone Administration Guide* for Unified Communications Manager for your Cisco Unified IP Phone model.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 12</td>
<td>Using a supported Cisco Unified IP Phone, establish the VPN connection.</td>
</tr>
</tbody>
</table>

Complete Cisco IOS Prerequisites

**Procedure**

**Step 1**

Install Cisco IOS Software version 15.1(2)T or later.

Feature Set/License: Universal (Data & Security & UC) for IOS ISR-G2

Feature Set/License: Advanced Security for IOS ISR

**Step 2**

Activate the SSL VPN License.

Configure Cisco IOS SSL VPN to Support IP Phones

**Procedure**

**Step 1**

Configure Cisco IOS locally.

a) Configure the Network Interface.

Example:

```
routing(config)# interface GigabitEthernet0/0
routert(config-if)# description "outside interface"
routert(config-if)# ip address 10.1.1.1 255.255.255.0
routert(config-if)# duplex auto
routert(config-if)# speed auto
routert(config-if)# shutdown
routert# show ip interface brief (shows interfaces summary)
```

b) Configure static and default routes by using this command:

```
routing(config)# ip route <dest_ip> <mask> <gateway_ip>
```

Example:

```
routing(config)# ip route 10.10.0.0 255.255.255.0 192.168.1.1
```

**Step 2**

Generate and register the CAPF certificate to authenticate the IP phones with an LSC.

**Step 3**

Import the CAPF certificate from Unified Communications Manager.

a) From the Cisco Unified OS Administration, choose Security > Certificate Management.
**Note** This location changes based on the Unified Communications Manager version.

b) Find the Cisco_Manufacturing_CA and CAPF certificates. Download the .pem file and save as .txt file.

c) Create trustpoint on the Cisco IOS software.

```
hostname(config)# crypto pki trustpoint trustpoint_name
hostname(config-ca-trustpoint)# enrollment terminal
hostname(config)# crypto pki authenticate trustpoint
```

When prompted for the base 64-encoded CA certificate, copy and paste the text in the downloaded .pem file along with the BEGIN and END lines. Repeat the procedure for the other certificates.

d) Generate the following Cisco IOS self-signed certificates and register them with Unified Communications Manager, or replace with a certificate that you import from a CA.

- Generate a self-signed certificate.

  ```
  Router> enable
  Router# configure terminal
  Router(config)# crypto key generate rsa general-keys label <name> <exportable -optional>
  Router(config)# crypto pki trustpoint <name> enrollment selfsigned
  Router(config-ca-trustpoint)# rsakeypair <name> 1024 1024
  Router(config-ca-trustpoint)# authorization username subjectname commonname
  Router(config-ca-trustpoint)# crypto pki enroll <name>
  Router(config-ca-trustpoint)# end
  ```

- Generate a self-signed certificate with Host-id check enabled on the VPN profile in Unified Communications Manager.

  Example:

  ```
  Router> enable
  Router# configure terminal
  Router(config)# crypto key generate rsa general-keys label <name> <exportable -optional>
  Router(config)# crypto pki trustpoint <name> enrollment selfsigned
  Router(config-ca-trustpoint)# fqdn <full domain name> Router(config-ca-trustpoint)# subject-name CN=<full domain name>, CN=<IP> Router(config-ca-trustpoint)# authorization username subjectname commonname
  Router(config-ca-trustpoint)# crypto pki enroll <name>
  Router(config-ca-trustpoint)# end
  ```

- Register the generated certificate with Unified Communications Manager.

  Example:

  ```
  Router(config)# crypto pki export <name> pem terminal
  ```

  Copy the text from the terminal and save it as a .pem file and upload it to the Unified Communications Manager using the Cisco Unified OS Administration.

**Step 4** Install AnyConnect on Cisco IOS.

Download the Anyconnect package from cisco.com and install to flash.

Example:

```
router(config)# webvpn install svc
flash:/webvpn/anyconnect-win-2.3.2016-k9.pkg
```

**Step 5** Configure the VPN feature.
To use the phone with both certificate and password authentication, create a user with the phone MAC address. Username matching is case sensitive. For example:

username CP-7975G-SEP001AE2BC16CB password k1kLGQIoxyCO4ti9 encrypted

Complete ASA Prerequisites for AnyConnect

Procedure

Step 1  Install ASA software (version 8.0.4 or later) and a compatible ASDM.
Step 2  Install a compatible AnyConnect package.
Step 3  Activate License.
   a) Check features of the current license using the following command:
   
   show activation-key detail
   
   b) If necessary, obtain a new license with additional SSL VPN sessions and enable the Linksys phone.
Step 4  Make sure that you configure a tunnel-group with a non-default URL as follows:

   tunnel-group phonevpn type remote-access
   tunnel-group phonevpn general-attribute
   address-pool vnpool
   tunnel-group phonevpn webvpn-attributes
   group-url https://172.18.254.172/phonevpn enable

Consider the following when configuring non-default URL:

   • If the IP address of the ASA has a public DNS entry, you can replace it with a Fully Qualified Domain Name (FQDN).
   
   • You can only use a single URL (FQDN or IP address) on the VPN gateway in Unified Communications Manager.
   
   • It is preferred to have the certificate CN or subject alternate name match the FQDN or IP address in the group-url.
   
   • If the ASA certificate CN or SAN does not match with the FQDN or IP address, uncheck the host ID check box in the Unified Communications Manager.

Configure ASA for VPN Client on IP Phone

Note

Replacing ASA certificates results in non-availability of Unified Communications Manager.
**Procedure**

**Step 1**  
Local configuration

a) Configure network interface.

Example:
```
ciscoasa(config)# interface Ethernet0/0
```
```
ciscoasa(config-if)# nameif outside
```
```
ciscoasa(config-if)# ip address 10.89.79.135 255.255.255.0
```
```
ciscoasa(config-if)# duplex auto
```
```
ciscoasa(config-if)# speed auto
```
```
ciscoasa(config-if)# no shutdown
```
```
ciscoasa(config-if)# show interface ip brief (shows interfaces summary)
```

b) Configure static routes and default routes.

```ciscoasa(config)# route <interface_name> <ip_address> <netmask> <gateway_ip>
```

Example:
```
ciscoasa(config)# route outside 0.0.0.0 0.0.0.0 10.89.79.129
```

c) Configure the DNS.

Example:
```
ciscoasa(config)# dns domain-lookup inside
```
```
ciscoasa(config)# dns server-group DefaultDNS
```
```
ciscoasa(config-dns-server-group)# name-server 10.1.1.5 192.168.1.67 209.165.201.6
```

**Step 2**  
Generate and register the necessary certificates for Unified Communications Manager and ASA.

Import the following certificates from the Unified Communications Manager.

- CallManager - Authenticating the Cisco UCM during TLS handshake (Only required for mixed-mode clusters).
- Cisco_Manufacturing_CA - Authenticating IP phones with a Manufacturer Installed Certificate (MIC).
- CAPF - Authenticating IP phones with an LSC.

To import these Unified Communications Manager certificates, do the following:

a) From the Cisco Unified OS Administration, choose **Security > Certificate Management**.

b) Locate the certificates Cisco_Manufacturing_CA and CAPF. Download the .pem file and save as a .txt file.

c) Create trustpoint on the ASA.

Example:
```
ciscoasa(config)# crypto ca trustpoint trustpoint_name
```
```
ciscoasa(ca-trustpoint)# enrollment terminal
```
```
ciscoasa(config)# crypto ca authenticate trustpoint_name
```

When prompted for base 64 encoded CA Certificate, copy-paste the text in the downloaded .pem file along with the BEGIN and END lines. Repeat the procedure for the other certificates.

d) Generate the following ASA self-signed certificates and register them with Unified Communications Manager, or replace with a certificate that you import from a CA.
• Generate a self-signed certificate.

Example:
```
ciscoasa> enable
ciscoasa# configure terminal
ciscoasa(config)# crypto key generate rsa general-keys label <name>
ciscoasa(config)# crypto ca trustpoint <name>
ciscoasa(config)# enrollment self
ciscoasa(config)# keypair <name>
ciscoasa(config)# crypto ca enroll <name>
ciscoasa(config)# end
```

• Generate a self-signed certificate with Host-id check enabled on the VPN profile in Unified Communications Manager.

Example:
```
ciscoasa> enable
ciscoasa# configure terminal
ciscoasa(config)# crypto key generate rsa general-keys label <name>
ciscoasa(config)# crypto ca trustpoint <name>
ciscoasa(config)# enrollment self
ciscoasa(config)# keypair <name>
ciscoasa(config)# crypto ca trustpoint <name>
ciscoasa(config)# enrollment self
```
```
ciscoasa(config)# fqdn <full domain name>
ciscoasa(config)# subject-name CN=<full domain name>,CN=<IP>
ciscoasa(config)# crypto ca enroll <name>
ciscoasa(config)# end
```

• Register the generated certificate with Unified Communications Manager.

Example:
```
ciscoasa(config)# crypto ca export <name> identity-certificate
```

Copy the text from the terminal and save it as a .pem file and upload it to Unified Communications Manager.

Step 3  Configure the VPN feature. You can use the Sample ASA configuration summary below to guide you with the configuration.

**Note**  To use the phone with both certificate and password authentication, create a user with the phone MAC address. Username matching is case sensitive. For example:
```
ciscoasa(config)# username CP-7975G-SEP001AE2BC16CB password k1kLGQIoxyCO4ti9 encrypted
```
```
ciscoasa(config)# username CP-7975G-SEP001AE2BC16CB attributes
```
```
ciscoasa(config-username)# vpn-group-policy GroupPhoneWebvpn
```
```
ciscoasa(config-username)# service-type remote-access
```

ASA Certificate Configuration

For more information on ASA certificate configuration, see Configure AnyConnect VPN Phone with Certificate Authentication on an ASA
Upload VPN Concentrator Certificates

Generate a certificate on the ASA when you set it up to support the VPN feature. Download the generated certificate to your PC or workstation and then upload it to Unified Communications Manager using the procedure in this section. Unified Communications Manager saves the certificate in the Phone-VPN-trust list.

The ASA sends this certificate during the SSL handshake, and the Cisco Unified IP Phone compares it against the values stored in the Phone-VPN-trust list.

The Cisco Unified IP Phone sends its Manufacturer Installed Certificate (MIC) by default. If you configure the CAPF service, the Cisco Unified IP Phone sends its Locally Significant Certificate (LSC).

To use device level certificate authentication, install the root MIC or CAPF certificate in the ASA, so that the Cisco Unified IP Phone are trusted.

To upload certificates to Unified Communications Manager, use the Cisco Unified OS Administration.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified OS Administration, choose <strong>Security &gt; Certificate Management</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Upload Certificate</strong>. The <strong>Upload Certificate</strong> dialog box appears.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Certificate Purpose</strong> drop-down list, choose <strong>Phone-VPN-trust</strong>.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Browse</strong> to choose the file that you want to upload.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Upload File</strong>.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Choose another file to upload or click <strong>Close</strong>.</td>
</tr>
</tbody>
</table>

For more information about **Certificate Management**, see **Administration Guide for Cisco Unified Communications Manager**.

---

Configure VPN Gateway

**Before you begin**

Ensure that you have configured VPN concentrators for each VPN gateway. After configuring the VPN concentrators, upload the VPN concentrator certificates. For more information, see **Upload VPN Concentrator Certificates**, on page 94.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Advanced Features &gt; VPN &gt; VPN Gateway</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Perform one of the following tasks:</td>
</tr>
<tr>
<td></td>
<td>a) Click <strong>Add New</strong> to configure new profile.</td>
</tr>
<tr>
<td></td>
<td>b) Click the <strong>Copy</strong> next to the VPN gateway that you want to copy.</td>
</tr>
<tr>
<td></td>
<td>c) Locate the appropriate VPN gateway and modify the settings to update an existing profile.</td>
</tr>
</tbody>
</table>
Step 3  Configure the fields in the **VPN Gateway Configuration** window. For more information, see **VPN Gateway Fields for VPN Client**, on page 95.

Step 4  Click **Save**.

## VPN Gateway Fields for VPN Client

*Table 10: VPN Gateway Fields for VPN Client*

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>VPN Gateway Name</td>
<td>Enter the name of the VPN gateway.</td>
</tr>
<tr>
<td>VPN Gateway Description</td>
<td>Enter a description of the VPN gateway.</td>
</tr>
</tbody>
</table>
| VPN Gateway URL               | Enter the URL for the main VPN concentrator in the gateway. You must configure the VPN concentrator with a group URL and use this URL as the gateway URL. For configuration information, refer to the documentation for the VPN concentrator, such as the following:
  * SSL VPN Client (SVC) on ASA with ASDM Configuration Example
| Note                          | If you do not assign a certificate for the gateway, the VPN client fails to connect to that concentrator.                                        |
| VPN Certificates in this Gateway | Use the up and down arrow keys to assign certificates to the gateway. If you do not assign a certificate for the gateway, the VPN client fails to connect to that concentrator. You can assign up to 10 certificates to a VPN gateway, and you must assign at least one certificate to each gateway. Only certificates that are associated with the Phone-VPN-trust role appear in the available VPN certificates list. |

## Configure VPN Group

*Procedure*

**Step 1**  From Cisco Unified CM Administration, choose **Advanced Features** > **VPN** > **VPN Group**.

**Step 2**  Perform one of the following tasks:
  a)  Click **Add New** to configure new profile.
  b)  Click **Copy** next to the VPN group that you want to copy an existing VPN group.
  c)  Locate the appropriate VPN group and modify the settings to update an existing profile.

**Step 3**  Configure the fields in the **VPN Group Configuration** window. For more information, see **VPN Gateway Fields for VPN Client**, on page 95 for the field description details.
Step 4 Click Save.

VPN Group Fields for VPN Client

<table>
<thead>
<tr>
<th>Field</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>VPN Group Name</td>
<td>Enter the name of the VPN group.</td>
</tr>
<tr>
<td>VPN Group Description</td>
<td>Enter a description of the VPN group.</td>
</tr>
<tr>
<td>All Available VPN Gateways</td>
<td>Scroll to see all available VPN gateways.</td>
</tr>
<tr>
<td>Selected VPN Gateways in this</td>
<td>Use the up and down arrow buttons to move available VPN gateways into and</td>
</tr>
<tr>
<td>VPN Group</td>
<td>out of this VPN group.</td>
</tr>
<tr>
<td></td>
<td>If the VPN client encounters critical error and cannot connect to a</td>
</tr>
<tr>
<td></td>
<td>particular VPN gateway, it will attempt to move to the next VPN</td>
</tr>
<tr>
<td></td>
<td>gateway in the list.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> You can add up to a maximum of three VPN gateways to a VPN group.</td>
</tr>
<tr>
<td></td>
<td>Also, the total number of certificates in the VPN group cannot exceed</td>
</tr>
<tr>
<td></td>
<td>10.</td>
</tr>
</tbody>
</table>

Configure VPN Profile

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Advanced Features > VPN > VPN Profile.

**Step 2** Perform one of the following tasks:

a) Click Add New to configure new profile.

b) Click Copy next to the VPN profile that you want to copy an existing profile.

c) To update an existing profile, specify the appropriate filters in the Find VPN Profile Where, click Find, and modify the settings.

**Step 3** Configure the fields in the VPN Profile Configuration window. For more information, see VPN Profile Fields for VPN Client, on page 97 for the field description details.

**Step 4** Click Save.
VPN Profile Fields for VPN Client

Table 12: VPN Profile Field Details

<table>
<thead>
<tr>
<th>Field</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>Enter a name for the VPN profile.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description for the VPN profile.</td>
</tr>
<tr>
<td>Enable Auto Network Detect</td>
<td>When you check this check box, the VPN client can only run when it detects that it is out of the corporate network. Default: Disabled.</td>
</tr>
<tr>
<td>MTU</td>
<td>Enter the size, in bytes, for the Maximum Transmission Unit (MTU). Default: 1290 bytes.</td>
</tr>
<tr>
<td>Fail to Connect</td>
<td>This field specifies the amount of time to wait for login or connect operations to complete while the system creates the VPN tunnel. Default: 30 seconds</td>
</tr>
<tr>
<td>Enable Host ID Check</td>
<td>When you check this check box, the gateway certificate subjectAltName or CN must match the URL to which the VPN client is connected. Default: Enabled</td>
</tr>
<tr>
<td>Client Authentication Method</td>
<td>From the drop-down list, choose the client authentication method:</td>
</tr>
<tr>
<td></td>
<td>• User and password</td>
</tr>
<tr>
<td></td>
<td>• Password only</td>
</tr>
<tr>
<td></td>
<td>• Certificate (LSC or MIC)</td>
</tr>
<tr>
<td>Enable Password Persistence</td>
<td>When you check this check box, a user password gets saved in the phone until either a failed log in attempt occurs, a user manually clears the password, or the phone resets or loses power.</td>
</tr>
</tbody>
</table>

Configure VPN Feature Parameters

Procedure

Step 1 From Cisco Unified CM Administration, choose Advanced Features > VPN > VPN Feature Configuration.
Step 2 Configure the fields in the VPN Feature Configuration window. For more information, see VPN Feature Parameters, on page 98.
Step 3 Click Save.
What to do next

Perform the following tasks:

- Upgrade the firmware for Cisco Unified IP Phones to a version that supports VPN. For more information about upgrading the firmware, see Cisco Unified IP Phone Administration Guide for your Cisco Unified IP Phone model.
- Using a supported Cisco Unified IP Phone, establish the VPN connection.

VPN Feature Parameters

Table 13: VPN Feature Parameters

<table>
<thead>
<tr>
<th>Field</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Auto Network Detect</td>
<td>When True, the VPN client can only run when it detects that it is out of the corporate network. Default: False</td>
</tr>
<tr>
<td>MTU</td>
<td>This field specifies the maximum transmission unit: Default: 1290 bytes Minimum: 256 bytes Maximum: 1406 bytes</td>
</tr>
<tr>
<td>Keep Alive</td>
<td>This field specifies the rate at which the system sends the keep alive message. Note If it is non zero and less than the value specified in Unified Communications Manager, the keep alive setting in the VPN concentrator overwrites this setting. Default: 60 seconds Minimum: 0 Maximum: 120 seconds</td>
</tr>
<tr>
<td>Fail to Connect</td>
<td>This field specifies the amount of time to wait for login or connect operations to complete while the system creates the VPN tunnel. Default: 30 seconds Minimum: 0 Maximum: 600 seconds</td>
</tr>
<tr>
<td>Client Authentication Method</td>
<td>From the drop-down list, choose the client authentication method: • User and password • Password only • Certificate (LSC or MIC) Default: User And Password</td>
</tr>
</tbody>
</table>
When True, a user password gets saved in the phone, if Reset button or "***" is used for reset. The password does not get saved and the phone prompts for credentials if the phone loses power or you initiate a factory reset.

Default: False

When True, the gateway certificate subjectAltName or CN must match the URL to which the VPN client is connected.

Default: True

**Add VPN Details to Common Phone Profile**

**Procedure**

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

**Step 2**
Click **Find** and choose common phone profile to which you want to add the VPN details.

**Step 3**
In the **VPN Information** section, choose the appropriate **VPN Group** and **VPN Profile**.

**Step 4**
Click **Save** and then **Apply Config**.

**Step 5**
Click **OK** in apply configuration window.
Add VPN Details to Common Phone Profile
PART IV

Monitoring and Recording

• Silent Monitoring, on page 103
• Recording, on page 111
Silent Monitoring Overview

Silent call monitoring allows a supervisor to eavesdrop on a phone conversation. The most common scenario is in a call center where a call agent is speaking with a customer. Call centers need to be able to guarantee the quality of customer service that an agent in a call center provides. With silent monitoring, the supervisor can hear both call participants, but neither of the call participants can hear the supervisor.

Silent monitoring can only be invoked by a CTI application through the JTAPI or TAPI interfaces. Many Cisco applications, such as Cisco Unified Contact Center Enterprise and Cisco Unified Contact Center Express have the ability to use silent monitoring. Any CTI application that monitors calls must have the corresponding monitoring privileges that are enabled for the application-user or end-user account.

Silent monitoring is call based. When a supervisor invokes a silent monitoring session, the following occurs:

- The supervisor selects a specific call to be monitored.
- The start-monitoring request from the application triggers the supervisor phone to go off hook and automatically triggers a monitoring call to the agent.
- The agent phone automatically answers the monitoring call. The monitoring call does not get presented to the agent.

Secure Silent Monitoring

You can also configure secure silent monitoring. Secure silent monitoring allows encrypted media (sRTP) calls to be monitored. Monitoring calls are always established using the highest level of security that is determined by the capabilities of the agent phone regardless of the security status of the call being observed. The highest level of security is maintained by exchanging the secure media key in any call between the customer, agent, and supervisor. Monitoring calls using secured media carries approximately 4000 bits per second of additional bandwidth overhead, same as standard secure media (sRTP) calls.
If the agent phone has encryption that is enabled, the supervisor phone must also have encryption enabled in order to allow secure silent monitoring. If the agent phone has encryption that is enabled, but the supervisor phone does not, the monitoring request fails.

**Whisper Coaching**

Unified Communications Manager also supports whisper coaching, a CTI enhancement on silent monitoring whereby a supervisor can speak to the agent while the monitoring session is underway without the customer hearing. Whisper coaching can only be initiated by a CTI application. If silent monitoring is already configured, then no additional configuration of Unified Communications Manager is required for whisper coaching.

**Silent Monitoring Prerequisites**

Silent monitoring can only be invoked by an external CTI application. Cisco applications such as Cisco Unified Contact Center Enterprise or Cisco Unified Contact Center Express can initiate silent monitoring sessions. For details, see the following:

- Cisco Unified Contact Center Enterprise—For details on how to set up silent monitoring in Cisco Unified Contact Center Enterprise, see Cisco Remote Silent Monitoring Installation and Administration Guide.
- Cisco Unified Contact Center Express—For details on how to set up silent monitoring in Cisco Unified Contact Center Express, see Cisco Unified CCX Administration Guide.

**Configure Silent Monitoring Task Flow**

This task flow describes the tasks that you must perform within Unified Communications Manager to allow CTI applications to use the monitoring feature.

**Before you begin**

- Determine which phones support silent monitoring by running a phone feature list report. For more information, *Generate a Phone Feature List, on page 1*

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | Perform one of the following procedures:  
- Enable Built in Bridge for Phones Clusterwide, on page 105  
- Enable Built in Bridge for a Phone, on page 105 | Turn on the Built in Bridge on agent phones. You can use a service parameter to configure the clusterwide default setting or you can enable the Built in Bridge for individual phones.  
**Note** The Built in Bridge setting on individual phones overrides the clusterwide default setting. |
| **Step 2** | Enable Monitoring Privileges for Supervisor, on page 106 | Add the supervisor to a group that allows silent monitoring. |
### Enable Built in Bridge for Phones Clusterwide

When you set the Built-in-Bridge clusterwide service parameter to enable, the Built-in-Bridge default setting for all phones in the cluster is changed to enabled. However, the Built-in-Bridge setting in the Phone Configuration window for individual phones overrides the clusterwide service parameter.

#### Procedure

1. From Cisco Unified CM Administration, choose **System > Service Parameters**.
2. From the **Server** drop-down list, choose the server on which the CallManager service is running.
3. From the **Service** drop-down list, choose **Cisco CallManager**.
4. Set the **Builtin Bridge Enable** service parameter to **On**.
5. Click **Save**.

### Enable Built in Bridge for a Phone

Use this procedure to enable the Built in Bridge on an individual phone. The Built in Bridge setting on an individual phone overrides the clusterwide service parameter.

#### Before you begin

Use a service parameter to set the Built in Bridge defaults for all phones in the cluster. For details, see **Enable Built in Bridge for Phones Clusterwide**, on page 105.

#### Procedure

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Find** to select the agent phone.
3. From the **Built in Bridge** drop-down list, choose one of the following options:
   - **On**—The Built in Bridge is enabled.
   - **Off**—The Built in Bridge is disabled.
Enable Monitoring Privileges for Supervisor

In order for a supervisor to be able to monitor agent conversations, the supervisor must be part of a group that allows monitoring.

Before you begin
Perform one of the following procedures to enable the Built in Bridge on agent phones:
- Enable Built in Bridge for Phones Clusterwide, on page 105
- Enable Built in Bridge for a Phone, on page 105

Procedure

Step 1 From Cisco Unified CM Administration, choose User Management > End User.
Step 2 Select the supervisor from the list of users.
Step 3 In the Permissions Information section, click Add to Access Control Group.
Step 4 Add the Standard CTI Allow Call Monitoring and Standard CTI Enabled user groups.
Step 5 Click Save.

Assign a Monitoring Calling Search Space

For monitoring to work, you must assign a Monitoring Calling Search Space to the supervisor phone line. The Monitoring Calling Search Space must include both the supervisor phone line and the agent phone line.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Find and select the supervisor phone.
The left navigation pane displays the available phone lines for the supervisor's phone.
Step 3 Perform the following steps for each of the supervisor's phone lines that are used for monitoring:
a) Click the phone line. The Directory Number Configuration window displays configuration information for that phone line.
b) From the Monitoring Calling Search Space drop-down list box, choose a calling search space that includes both the supervisor phone line and the agent phone line.
c) Click Save.
Configure Silent Monitoring Notification Tones

In certain jurisdictions, a notification tone must be played to either the agent, the customer, or both, that indicates that the call is being monitored. By default, Unified Communications Manager does not play notification tones. You must configure a service parameter to allow notification tones.

Procedure

| Step 1 | From Cisco Unified CM Administration, choose System > Service Parameters. |
| Step 2 | From the Server drop-down list box, choose the server on which the CallManager service is running. |
| Step 3 | From the Service drop-down list box, choose Cisco CallManager. |
| Step 4 | Configure values for the following service parameters: |
|       | • If you want to play a notification tone to the agent, change the value of the Play Monitoring Notification Tone To Observed Target service parameter to True. |
|       | • If you want to play a notification tone to the customer, change the value of the Play Monitoring Notification Tone To Observed Connected Parties service parameter to True. |
| Step 5 | Click Save. |
| Step 6 | Reset the agent phone, if you changed the service parameter configuration. |

Configure Secure Silent Monitoring

To configure secure silent monitoring using sRTP, you must configure phone security profiles that include encryption and apply them to the supervisor phone and to any agent phones that are being monitored.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure an Encrypted Phone Security Profile, on page 107</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Assign Security Profile to Phone, on page 108</td>
</tr>
</tbody>
</table>

Configure an Encrypted Phone Security Profile

To configure secure silent monitoring, you must configure the phone security profile for your supervisor phone and any agent phones to specify Encrypted as the Device Security Mode.

Procedure

| Step 1 | From Cisco Unified CM Administration, choose System > Security > Phone Security Profile. |
| Step 2 | Perform either of the following steps: |
• Click Add New to create a new phone security profile.
• Click Find and select an existing phone security profile.

Step 3  If you have created a new phone security profile, select your phone model from the Phone Security Profile Type drop-down list.

Step 4  Enter a Name for the Phone Security Profile.

Step 5  From the Device Security Mode drop-down list, choose Encrypted.

Step 6  Click Save.

Step 7  Repeat the above steps to configure phone security profiles for your supervisor phone and any agent phones.

Assign Security Profile to Phone

Perform the following steps to assign a phone security profile to a phone. For secure silent monitoring to work, you must assign the phone security profile to both the agent phone and the supervisor phone.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Device &gt; Phone.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Find and select the agent phone on which you want to configure a phone security profile.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Device Security Profile drop-down list, choose the phone security profile that you have set up.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Repeat the previous steps for the supervisor phone.</td>
</tr>
</tbody>
</table>

Cisco Unified Contact Center Express Configuration Task Flow


Procedure

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Create the monitoring supervisor.</td>
<td>Use the User Management web page to assign supervisor privileges to a user.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign resources and contact service queues to a supervisor.</td>
<td>Use the Remote Monitor Configuration web page to assign a supervisor a list of resources and CSQs they are allowed to monitor.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the remote monitoring application.</td>
<td>The Remote Monitoring application comes with Unified CCX Premium systems. You must configure Remote Monitoring applications when you want to use the Remote Monitoring feature.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td>View contact service queues for remote monitoring.</td>
<td>View contact service queue IDs from the Unified CCX Supervisor web page.</td>
<td></td>
</tr>
</tbody>
</table>

## Silent Monitoring Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call preservation</td>
<td>If the agent call that is being monitored goes to call preservation, Unified Communications Manager also puts the monitoring call into call preservation mode.</td>
</tr>
<tr>
<td>Transfer of secure monitoring call</td>
<td>Unified Communications Manager supports transferring a secure monitoring session so long as the destination supervisor device exceeds the security capabilities of the agent that is being monitored.</td>
</tr>
<tr>
<td>Recording Tones</td>
<td>Recording Tones take precedence over Monitoring Tones for calls that are both recorded and monitored. If a call is recorded and monitored, only the recording tone plays.</td>
</tr>
<tr>
<td>Secure Tones</td>
<td>If Secure Tones are configured and the call is secured, the secure tone plays to both call participants at the outset of the call irrespective of whether Monitoring Tones are configured. If Secure Tones and Monitoring Tones are both configured, the secure tone plays once, followed by the monitoring tones. If Secure Tones, Monitoring Tones, and Recording Tones are all configured, and the call is recorded and monitored, the secure tone plays once followed by the recording tone. The monitoring tone does not play.</td>
</tr>
</tbody>
</table>

## Silent Monitoring Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Barge</td>
<td>Unified Communications Manager does not support barge with silent monitoring. If an agent call is being monitored, the barge-in call from a shared line fails. If the agent call has already been barged, the monitoring call fails.</td>
</tr>
<tr>
<td>Transfer of Secure Silent Monitoring over an intercluster trunk</td>
<td>Unified Communications Manager does not support transferring Secure Silent Monitoring calls over an intercluster trunk.</td>
</tr>
</tbody>
</table>
CHAPTER 12

Recording

• Recording Overview, on page 111
• Recording Prerequisites, on page 114
• Recording Configuration Task Flow, on page 115
• Recording Call Flow Examples, on page 124
• Recording Interactions and Restrictions, on page 124

Recording Overview

Call recording is a Unified Communications Manager feature that enables a recording server to archive agent conversations. Call recording is one of the essential features in call centers, financial institutions and other enterprises. The call recording feature sends copies of the agent and the end-user media streams to the recording server over a SIP trunk. Each media stream is sent separately in an effort to best support a wide range of voice analytic applications.

Unified Communications Manager offers IP phone-based or network-based recording.

• In IP phone based recording, recording media is sourced from the phone. The phone forks two media streams to the recording server.

• In network-based recording, recording media can be sourced from either the phone or the gateway. When you implement network-based recording, the gateway in your network must connect to Unified Communications Manager over a SIP trunk.

Unified Communications Manager supports call recording in both single cluster and multi-cluster environments and offers three different recording modes:

• Automatic Silent Recording—Automatic silent recording records all calls on a line appearance automatically. Unified Communications Manager invokes the recording session automatically with no visual indication on the phone that an active recording session is established.

• Selective Silent Recording—A supervisor can start or stop the recording session via CTI-enabled desktop. Alternatively, a recording server can invoke the session based on predefined business rules and events. There is no visual indication on the phone that an active recording session is established.

• Selective User Call Recording—An agent can choose which calls to record. The agent invokes the recording session through CTI-enabled desktop, or by a softkey or programmable line key. When selective user call recording is used, the Cisco IP phone displays recording session status messages.
Unified Communications Manager supports recording to a single recording server and can be deployed with CUBE as media proxy to record to more than one recording server.

- In multi-fork recording, Unified Communications Manager is connected to CUBE Media Proxy server through SIP trunk. The CUBE Media Proxy server receives two media streams from phone or gateway and it forks the media streams to one or more recording servers simultaneously.

- In recording to single recording servers, Unified Communications Manager is directly connected to recording server through SIP trunk. The phone or the gateway forks two media streams to the recording server.

## Multi-Fork Recording

Unified Communications Manager supports simultaneous, multiple stream recordings through Cisco Unified Border Element (CUBE) as Media Proxy. In Multi-fork recording, the recording stream is sent to a CUBE Media Proxy server, which relays the media stream to up to five recording servers simultaneously. This is supported for both phone-based and network-based recording, and for both automatic and selective recording.

The multi-forking feature provides the following benefits:

- Adds redundancy and failover to your recording deployment.
- Provides additional media streams for speech analysis and monitoring.
- Helps organization, such as financial industry, to be compliant to MiFID requirements, that mandate recording of customer calls to multiple recording servers for redundancy.

When you implement multi-fork recording, you must configure the CUBE Media Proxy server in your network which connects to Unified Communications Manager over a SIP trunk.

For more information on CUBE Media Proxy, see Cisco Unified Border Element Configuration Guide.

---

**Note**

Connection from Unified Communications Manager to CUBE Media Proxy server over a SIP trunk must be configured with Early Offer.

The following example illustrates the phone-based recording of multi-fork recording through CUBE Media Proxy.

*Figure 4: Phone-based recording*
The following example illustrates the network-based recording of multi-fork recording through CUBE Media Proxy.

**Figure 5: Network-based recording**

For more information on method summary, see "Cisco Device-Specific Extensions" section of Cisco Unified JTAPI Developers Guide for Cisco Unified Communications Manager Release 12.5(1).

**Supported Platforms**

Multi-fork recording through CUBE Media Proxy server is supported on the following Cisco Router Platforms running on Cisco IOS XE Gibraltar Release 16.10.1:

- Cisco 4000 Series-Integrated Services Routers (ISR G3 - ISR4331, ISR4351, ISR4431, ISR4451).
- Cisco Cloud Services Routers (CSR1000V series).

**Restrictions for Multi-fork recording through CUBE Media Proxy**

Multi-fork recording through CUBE Media Proxy server does not support the following:

- Video recording.
- Secure media (SRTP) forking of non secure calls.
- SRTP fall back.
- Midcall block.

**Recording Media Source Selection**

When you configure network-based recording, you must configure either the phone or the gateway as your preferred source of recording media for the agent phone line. However, depending on your deployment, Unified Communications Manager may not select your preferred choice as the recording media source. The following table displays the logic Unified Communications Manager uses to select the recording media source.
### Table 14: Recording Media Source Selection

<table>
<thead>
<tr>
<th>Preferred Media Source</th>
<th>Media Type</th>
<th>Gateway in call path?</th>
<th>Selected Media Source</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway</td>
<td>Unsecure (RTP)</td>
<td>Yes</td>
<td>Gateway</td>
</tr>
<tr>
<td></td>
<td></td>
<td>No</td>
<td>Phone</td>
</tr>
<tr>
<td></td>
<td>Secure (sRTP)</td>
<td>Yes</td>
<td>Phone</td>
</tr>
<tr>
<td></td>
<td></td>
<td>No</td>
<td>Phone</td>
</tr>
<tr>
<td>Phone</td>
<td>Unsecure (RTP)</td>
<td>Yes</td>
<td>Phone</td>
</tr>
<tr>
<td></td>
<td></td>
<td>No</td>
<td>Phone</td>
</tr>
<tr>
<td></td>
<td>Secure (sRTP)</td>
<td>Yes</td>
<td>Phone</td>
</tr>
<tr>
<td></td>
<td></td>
<td>No</td>
<td>Phone</td>
</tr>
</tbody>
</table>

### Alternate Recording Media Source if the First Choice is Unavailable

If the recording media source that Unified Communications Manager selects is unavailable, Unified Communications Manager attempts to use an alternate source. The following table shows the logic Unified Communications Manager uses to select an alternate source for recording media.

### Table 15: Alternate Recording Media Source if First Choice is Unavailable

<table>
<thead>
<tr>
<th>Selected Media Source</th>
<th>Gateway Preferred</th>
<th>Phone Preferred</th>
</tr>
</thead>
<tbody>
<tr>
<td>First attempt</td>
<td>First gateway in call path</td>
<td>Phone</td>
</tr>
<tr>
<td>Second attempt</td>
<td>Last gateway in call path</td>
<td>First gateway in call path</td>
</tr>
<tr>
<td>Third attempt</td>
<td>Phone</td>
<td>Last gateway in call path</td>
</tr>
</tbody>
</table>

### Recording Prerequisites

- **Cisco Unified IP Phone support**—To view a list of the Cisco Unified IP Phone that support recording, log in to Cisco Unified Reporting and run the Unified CM Phone Feature List report, selecting **Record** as the feature. For a detailed procedure, see Generate a Phone Feature List, on page 1.

- **Gateway support**—For details on which gateways support recording, see https://developer.cisco.com/web/sip/wiki/-/wiki/Main/Unified+CM+Recording+Gateway+Requirements.

- **If you are configuring multiple-stream recording**, deploy and configure a CUBE Media Proxy. For details, see the section **CUBE Media Proxy** in the Cisco Unified Border Element Configuration Guide.
Recording Configuration Task Flow

Before you begin

<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>Create a Recording Profile, on page 115</td>
<td>Create a recording profile.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure SIP Profile for Recording, on page 116</td>
<td>Optional. Configure the SIP Profile if you want to deliver the Conference Bridge Identifier to the recorder.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure SIP Trunks for Recording, on page 116</td>
<td>Configure the recorder server or CUBE Media Proxy as a SIP trunk device.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure Route Pattern for Recording, on page 117</td>
<td>Create a route pattern that routes to the recorder server or CUBE Media Proxy.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure Agent Phone Line for Recording, on page 117</td>
<td>Configure the agent phone line for recording.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Enable the built in bridge for your agent phones. Perform one of the following tasks to enable the built-in-bridge for recording:</td>
<td>To use the agent phone as the recording media source you must enable the phone's built in bridge for recording. You can use a service parameter to set the built in bridge defaults across the cluster, or enable the built in bridge on an individual phone.</td>
</tr>
<tr>
<td></td>
<td>• Enable Built in Bridge for Cluster, on page 118</td>
<td>Note The Built in Bridge setting on individual phones overrides the clusterwide defaults.</td>
</tr>
<tr>
<td></td>
<td>• Enable Built in Bridge for a Phone, on page 118</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Enable Gateway for Recording, on page 119</td>
<td>Configure Unified Communications services on the gateway.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Configure Recording Notification Tones, on page 119</td>
<td>Configure whether you want a notification tone to play when calls are recorded.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>Perform one of the following procedures, depending on whether your phone uses feature buttons or softkeys:</td>
<td>Configure a Record feature button or softkey for your phone.</td>
</tr>
<tr>
<td></td>
<td>• Configure a Record Feature Button, on page 120</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure a Record Softkey, on page 121</td>
<td></td>
</tr>
</tbody>
</table>

Create a Recording Profile

Use this procedure to create a recording profile.
**Configure SIP Profile for Recording**

Use this procedure to deliver the conference bridge identifier to the recorder and configure the SIP Profile.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose Device > Device Settings > Recording Profile.
- **Step 2** Click Add New.
- **Step 3** In the Name field, enter a name for your recording profile.
- **Step 4** In the Recording Calling Search Space field, select the calling search space that contains the partition with the route pattern that is configured for the recording server.
- **Step 5** In the Recording Destination Address field, enter the directory number or the URL of the recording server or the URL of the CUBE Media Proxy server.
- **Step 6** Click Save.

**Configure SIP Trunks for Recording**

Use this procedure to assign the recording server information in the SIP Trunk Configuration window.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.
- **Step 2** Select the SIP profile that you want to use for your network.
- **Step 3** Set a value for the Early Offer Support for Voice and Video calls field. SIP trunk from Unified Communications Manager to CUBE Media Proxy server must be enabled for an Early Offer support and configuration options are Best Effort (no MTP inserted) and Mandatory (insert MTP if needed).
- **Step 4** Check the Deliver Conference Bridge Identifier check box.
- **Step 5** Click Save.
Step 5  Click Next.

Step 6  In the Destination Address field of the SIP Information pane, enter an IP address, fully qualified domain name, or DNS SRV of the recording server or CUBE Media proxy.

Step 7  From the SIP Profile drop-down list in the SIP Information pane, choose the SIP profile that you want to use in your network.

Step 8  From the Recording Information pane, select one of the following options:

- None—This trunk is not used for recording.
- This trunk connects to a recording-enabled gateway.
- This trunk connects to other clusters with recording-enabled gateways.

Step 9  Click Save.

Note  SIP trunk from Unified Communications Manager to Media Proxy must be enabled for Early Offer support in the SIP Profile that is used for this trunk. The configuration options are Mandatory (insert MTP if needed) and Best Effort (no MTP inserted).

---

Configure Route Pattern for Recording

Use this procedure to describe the route pattern configurations that are specific to recorders. You must configure a route pattern that routes to the recording server or CUBE Media Proxy server.

Procedure

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

Step 2  Click Add New to create a new route pattern.

Step 3  Complete the fields in the Route Pattern Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 4  For call recording, complete the following fields:

- Pattern—Enter a pattern that matches the recording destination address from the recording profile.
- Gateway/Route List—Choose the SIP trunk or route list that points to the recording server.

Step 5  Click Save.

---

Configure Agent Phone Line for Recording

Use this procedure to configure the agent phone line for recording.

Procedure

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

Step 2  Click Find.
Step 3 Select the agent's phone.
Step 4 In the left Association pane, click the phone line for the agent to view the settings.
Step 5 From the Recording Option drop-down list, choose one of the following options:
   • Call Recording Disabled—Calls on this phone line are not recorded.
   • Automatic Call Recording Enabled—All calls on this phone line are recorded.
   • Selective Call Recording Enabled—Only selected calls on this phone line are recorded.
Step 6 From the Recording Profile drop-down list, choose the recording profile that is configured for the agent.
Step 7 From the Recording Media Source drop-down list, choose whether you want to use the gateway or the phone as the preferred source of recording media.
Step 8 Set the Busy Trigger field to a minimum of 3 if you also have Multilevel Precedence and Preemption (MLPP) configured.
Step 9 Click Save.

Enable Built in Bridge for Cluster

Use this procedure to enable the phone's built in bridge for recording to use the agent phone as the recording media source.

When you set the Built-in-Bridge clusterwide service parameter to enable, the Built-in-Bridge default setting for all phones in the cluster is changed to enabled. However, the Built-in-Bridge setting in the Phone Configuration window for an individual phone overrides the clusterwide service parameter setting if the default option is not selected for that phone.

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 CallManager service is running.
Step 3 From the Service drop-down list, choose Cisco CallManager.
Step 4 Set the Builtin Bridge Enable service parameter to On.
Step 5 Click Save.

Enable Built in Bridge for a Phone

Use this procedure to enable the Built in Bridge for an individual phone. If the default option is not selected, the Built in Bridge setting in the Phone Configuration window overrides the clusterwide service parameter.

Optionally, use a service parameter to set the Built in Bridge defaults across the cluster. For more information, see Enable Built in Bridge for Cluster , on page 118.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Click **Find** to select the agent phone.

Step 3  From the **Built in Bridge** drop-down list, choose one of the following options:

- **On** — The Built in Bridge is enabled.
- **Off** — The Built in Bridge is disabled.
- **Default** — The setting of the clusterwide **Built in Bridge Enable** service parameter is used.

Step 4  Click **Save**.

---

### Enable Gateway for Recording

Use this procedure to configure the gateway for recording. You must enable Unified Communications Gateway Services. The following task flow contains the high-level process to enable Unified Communications Gateway Services.

**Procedure**

**Step 1**  Configure Unified Communications Manager IOS Services on the Device.

**Step 2**  Configure the XMF Provider.

**Step 3**  Verify Unified Communications Gateway Services.

For detailed configuration steps, including examples, refer to the Cisco Unified Communications Gateway Services chapter for either of the following documents:

- For more information, see ASR routers **Cisco Unified Border Element (Enterprise) Protocol-Independent Features and Setup Configuration Guide. Cisco IOS XE Release 35**.

- For more information, see ISR routers **Cisco Unified Border Element Protocol-Independent Features and Setup Configuration Guide, Cisco IOS Release 15M&T**.

### Configure Recording Notification Tones

Use this procedure to configure notification tone to play when calls are recorded. For legal compliance, an explicit notification in the form of a periodic tone can be made audible to the agent, the caller, or both, to indicate that a recording session is in progress. This tone can also be disabled.

**Note**

Recording tone settings override monitoring tone settings when both are enabled for the same call.

**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.
Configure a Record Feature Button

Use this procedure to assign the Record feature button to your phone if your phone uses feature buttons.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure a Phone Button Template for Recording, on page 120</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Phone Button Template with a Phone, on page 121</td>
</tr>
</tbody>
</table>

Configure a Phone Button Template for Recording

Use this procedure to create a phone button template that includes the Record feature button.

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:
   • If you modified a template that is already associated with devices, click Apply Config to restart the devices.
• If you created a new softkey template, associate the template with the devices and then restart them.

### Associate a Phone Button Template with a Phone

Use this procedure to associate the phone button template that you created for the Record button of the phone.

**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 <strong>Device &gt; Phone</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> to display the list of configured phones.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose the phone to which you want to add the phone button template.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Phone Button Template</strong> drop-down list, choose the phone button template that contains the new feature button.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>. A dialog box is displayed with a message to press <strong>Reset</strong> to update the phone settings.</td>
<td></td>
</tr>
</tbody>
</table>

### Configure a Record Softkey

Use this procedure to add a Record softkey to the phone, if your phone uses softkeys. The Record softkey is only available in the Connected call state for the Cisco Chaperone Phone with Feature Hardkeys template.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Configure a Softkey Template for Recording, on page 121</th>
<th>Configure a softkey template that includes the Record softkey.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Perform one of the following procedure:</td>
<td>Associate the softkey template to a phone directly, or to a Common Device Configuration. You can then associate the Common Device Configuration to a group of phones.</td>
</tr>
<tr>
<td></td>
<td>• Associate a Softkey Template with a Phone, on page 122</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Associate a Softkey Template with a Common Device Configuration, on page 123</td>
<td></td>
</tr>
</tbody>
</table>

### Configure a Softkey Template for Recording

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Device Settings &gt; Softkey Template</strong>.</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Perform the following steps to create a new softkey template; otherwise, proceed to the next step.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>a) Click <strong>Add New</strong>.</td>
<td></td>
</tr>
</tbody>
</table>
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:

- If you modified a template that is already associated with devices, click **Apply Config** 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.

---

### Associate a Softkey Template with a Phone

Use this procedure to assign the Record softkey to the phone by associating the softkey template that includes the Record softkey directly to a phone.

**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.
Associate a Softkey Template with a Common Device Configuration

Use this procedure to add a Record softkey to the phone by associating the softkey template to a Common Device Configuration.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add a Softkey Template to the Common Device Configuration, on page 123</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Add Common Device Configuration to Phone, on page 123</td>
</tr>
</tbody>
</table>

Add a Softkey Template to the 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.

Add Common Device Configuration to 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.
**Recording Call Flow Examples**

For call flow examples for both network-based call recording and IP phone-based call recording use cases, refer to *Call Recording Examples for Network-Based and Phone-Based Recording* at the following URL:


**Recording Interactions and Restrictions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions and Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Monitoring Tones</td>
<td>Recording Tones take precedence over Monitoring Tones for calls that are both recorded and monitored. If both are configured, and a call is both recorded and monitored, only the recording tone plays.</td>
</tr>
<tr>
<td>Multilevel Precedence and Preemption</td>
<td>If you also have Multilevel Precedence and Preemption (MLPP) configured, the <strong>Busy Trigger</strong> setting on the agent phone line that you are recording must be set to a minimum of 3.</td>
</tr>
</tbody>
</table>
| Secure Tones               | If Secure Tones are configured, the secure tone plays to both call participants at the outset of a secure call, irrespective of whether Recording Tones are configured.  
                            | If Secure Tones and Recording Tones are both configured and the call is secure, the secure tone plays once at the outset of the call followed by the recording tone.  
                            | If Secure Tones, Recording Tones, and Monitoring Tones are all configured, and the call is secured, recorded, and monitored, the secure tone plays once followed by the recording tone. The monitoring tone does not play. |
| Customer Voice Portal       | Agent - customer calls that are routed through the Customer Voice Portal may be recorded using the agent phone as the recording source.                                  |
| SIP Proxy Servers           | If you are using the gateway as your recording source, you cannot place SIP proxy servers between Unified Communications Manager and the gateway.                                            |
| Busy Hour Call Completion Rate | Each recording session adds two calls to the Busy Hour Call Completion (BHCC) rate with a minimal impact on CTI resources.                                   |
### Monitoring and Recording

#### Recording Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions and Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Selective Recording with Media Sense</td>
<td>When Selective Recording is configured, the Media Sense server does not record the consult call during a transfer. For example, if a call between an agent and a customer is being recorded and the agent initiates a transfer to a second agent, the consult call that takes place between the two agents, prior to the call being transferred, is not recorded. To ensure that the consult call is recorded, the agent must press the ‘Record’ softkey when the consult call starts.</td>
</tr>
<tr>
<td>Recording on authenticated phones</td>
<td>To record a call for authenticated phones, On the Cisco Unified CM Service Parameter page, set the <strong>Authenticated Phone Recording</strong> field to <strong>Allow Recording</strong>. The default value is <strong>Do Not Allow Recording</strong>. Unified Communications Manager allows call recording for authenticated phones while using non secure recorder. In case of secure recorder, recording is allowed only if the recorder supports Secure Real-Time Transport protocol (SRTP) fallback.</td>
</tr>
<tr>
<td>Codec locking for auto recording calls in select and join conference</td>
<td>Skinny Client Control Protocol (SCCP) phone adversities one single codec when recording is enabled and there is a select and join conference performed in Unified Communications Manager.</td>
</tr>
</tbody>
</table>

**Recording Calls Do Not Survive Agent Hold**

Recording calls get torn down when the agent puts the call on hold, and they get reestablished when the agent resumes the call.

*Figure 6: Recording Calls Do Not Survive Agent Hold*
Recording Interactions and Restrictions
PART V

Call Center Features

• Agent Greeting, on page 129
• Auto-Attendant, on page 133
• Manager Assistant, on page 141
Agent Greeting Overview

Agent Greeting enables Unified Communications Manager to automatically play a prerecorded announcement following a successful media connection to the agent device. Agent Greeting is audible for the agent and the customer.

The process of recording a greeting is similar to recording a message for voicemail. Depending on how your contact center is set up, you can record different greetings that play for different types of callers (for example, an English greeting for English speakers or an Italian greeting for Italian speakers).

By default, agent greeting is enabled when you log in to your agent desktop but you can turn it off and on as necessary.

Agent Greeting Prerequisites

- Install Cisco Unified Contact Center Enterprise. See Cisco Unified Contact Center Enterprise Installation and Upgrade Guide.
- Ensure that you enable Built In Bridge. To view the details, see Configure Built In Bridge, on page 131.

Agent Greeting Configuration Task Flow

Agent Greeting configuration tasks are completed in Cisco Unified Contact Center Enterprise (Unified CCE) and Cisco Unified Customer Voice Portal (Unified CVP). To view detailed steps for the following tasks, see the Agent Greeting section in the Cisco Unified Contact Center Enterprise Features Guide.
Before you begin

- Review Agent Greeting Prerequisites, on page 129

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 1 | Configure a media server for Agent Greeting.  
• Configure a server to act as a media server.  
• Add the media server in Unified CVP.  
• Configure the media server to write files. | Agent Greeting uses the Unified CVP media server to store and serve prompt and greeting files. |
| Step 2 | Republish .tcl scripts to Voice Extensible Markup Language (VXML) Gateway. | The .tcl script files that ship with Unified CVP Release 9.0(1) include updates to support Agent Greeting. You must republish these updated files to your VXML Gateway. Republishing scripts to the VXML Gateways is a standard task in Unified CVP upgrades. If you did not upgrade Unified CVP and republish the scripts, you must republish the scripts before you can use Agent Greeting. |
| Step 3 | Set the cache size on the VXML Gateway. | To ensure adequate performance, set the size of the cache on the VXML Gateway to the maximum allowed. The maximum size is 100 megabytes; the default is 15 kilobytes. Failure to set the VXML Gateway cache to its maximum can result in slowed performance to increased traffic to the media server. |
| Step 4 | Create voice prompts to record greetings. | Create audio files for each of the voice prompts that agents hear as they record a greeting. |
| Step 5 | Configure call types. | Complete to record and play agent greetings. |
| Step 6 | Configure a dialed number. | Complete to record and play agent greetings. |
| Step 7 | Schedule the script. | |
| Step 8 | Define network VRU scripts. | For Agent Greeting record and play scripts to interact with Unified CVP, Network VRU scripts are required. |
| Step 9 | (Optional) Import sample Agent Greeting scripts. | |
| Step 10 | Modify the Unified CCE call routing scripts. | Modify the Unified CCE call routing scripts to use the Play Agent Greeting script. |
Configure Built In Bridge

The Built in Bridge field setting in the Phone Configuration window for an individual phone overrides the setting for the Builtin Bridge Enable clusterwide service parameter.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Find to select the agent phone.
Step 3 From the Built in Bridge drop-down list, choose one of the following options:
  • On—The Built in Bridge is enabled.
  • Off—The Built in Bridge is disabled.
  • Default—The setting of the clusterwide Builtin Bridge Enable service parameter is used.
Step 4 Click Save.

Agent Greeting Troubleshooting

For information about how to troubleshoot Agent Greeting issues, see “Troubleshooting Agent Greeting” chapter in the Agent Greeting and Whisper Announcement Feature Guide for Cisco Unified Contact Center Enterprise guide.
Agent Greeting Troubleshooting
Auto-Attendant Overview

Auto-Attendant allows callers to locate people in your organization without talking to a receptionist. You can customize the prompts that are played for the caller.

Auto-Attendant works with Unified Communications Manager to receive calls on specific telephone extensions. The software interacts with the caller and allows the caller to search for and select the extension of the party (in your organization) that the caller is trying to reach.

Auto-Attendant provides the following functions:

- Answers a call
- Plays a user-configurable welcome prompt
- Plays a main menu prompt that asks the caller to perform one of three actions:
  - Press 0 for the operator
  - Press 1 to enter an extension number
  - Press 2 to spell by name

If the caller chooses to spell by name (by pressing 2), the system compares the letters that are entered with the names that are configured to the available extensions. One of the following results can occur:

- If a match exists, the system announces a transfer to the matched user and waits for up to 2 seconds for the caller to press any Dual Tone Multifrequency (DTMF) key to stop the transfer. If the caller does not stop the transfer, the system performs an explicit confirmation: it prompts the user for confirmation of the name and transfers the call to the primary extension of that user.
- If more than one match occurs, the system prompts the caller to choose the correct extension.
- If too many matches occur, the system prompts the caller to enter more characters.
• If no match occurs, that is, if the user presses wrong options, the system prompts that the user pressed the wrong options and prompts the user to press the correct options.

• When the caller specifies the destination, the system transfers the call.

• If the line is busy or not in service, the system informs the caller accordingly and replays the main menu prompt.

Auto-Attendant solution can be deployed in three different ways as follows using different Cisco products that can provide interactive voice response functionality.

• Auto-Attendant using Cisco Unity Connection (CUC); the most widely used Auto-Attendant solution configuration by customers

• Auto-Attendant using Cisco Unified Contact Center Express (Unified CCX)

• Auto-Attendant using Cisco Unity Express (CUE)

**Cisco Unity Connection Configuration**

The Cisco Unity Connection server provides Automated-Attendant functionality for both external and internal callers. An Auto-Attendant allows callers to be automatically transferred to an extension without the intervention of an operator or receptionist.

Auto-Attendants offer a menu system; it may also allow a caller to reach a live operator by dialing a number, usually “0”. Multiple Auto-Attendants may be implemented to support individual site locations. Within Cisco Unity Connection, an Auto-Attendant is a customized application tree structure that is built by creating and linking multiple Call Handlers together. The Auto-Attendant is defined by entry and exit points, and intermediate routing decisions based on the callers DTMF input choices.

For more information about Auto-Attendant default behavior and examples, see System Administration Guide for Cisco Unity Connection.

**Cisco Unity Connection Configuration Task Flow**

You can use this task flow to configure auto-attendant using Cisco Unity Connection:

<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 CTI Route Point, on page 135</td>
<td>Perform this task on the Cisco Unified CM Administration. Create a CTI Route Point which maps to the Direct-Inward Dial (DID) number of the company (board number).</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Auto-Attendant System Call Handler, on page 136</td>
<td>Call handlers answer calls, greet callers with recorded prompts, provide callers with information and options, route calls, and take messages.</td>
</tr>
</tbody>
</table>
### Purpose

Command or Action | Purpose
---|---
| **Note** | You can customize the greeting for the AutoAttendant Call Handler by choosing **Edit > Greetings**. For more information about customizing greetings, see **System Administration Guide for Cisco Unity Connection**.
| **Step 3** | **Configure Caller Input Option, on page 136**
Caller input option enables you to designate a single digit to represent a user extension, alternate contact number, call handler, interview handler, or directory handler. The caller presses a single key during a call handler greeting instead of entering the full extension, and Cisco Unity Connection responds accordingly. Several different keys configured as caller input options offers the callers a menu of choices in the call handler greeting.
| **Step 4** | **Configure Extension for Operator Call Handler, on page 137**
Configure an extension for the operator to allow callers to speak to an operator during a call handler greeting.
| **Step 5** | **Modify Standard Call Transfer Rule for Operator, on page 137**
Modify the Standard Call Transfer Rule to enable the call to be transferred to the operator when the caller presses 0 to speak to an operator.
| **Step 6** | **Update Default System Transfer Restriction Table, on page 137**
Update the Default System Transfer restriction table. The Default System Transfer restriction table restricts numbers that can be used for Caller system transfers, which allow unidentified callers to transfer to a number that they specify.

### Configure CTI Route Point

#### Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > CTI Route Point**.

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

**Step 3** In the **Device Name** field, enter a device name for the route point.

**Step 4** From the **Device Pool** drop-down list, choose **Default**.

**Step 5** Click **Save**. The **Add successful** message is displayed.

**Step 6** From the **Association** area, click **Line [1] - Add a new DN**. The **Directory Number Configuration** window is displayed.

**Step 7** In the **Directory Number** field, enter the directory number that matches with the DID of the company.

**Step 8** From the **Route Partition** drop-down list, choose the required route partition.
Configure Auto-Attendant System Call Handler

Procedure

Step 1 From Cisco Unity Connection Administration, from the Cisco Unity Connection tree on the left, navigate to Call Management and choose System Call Handlers.

Step 2 Click Add New.
The New Call Handler window is displayed.

Step 3 In the Display Name field, enter AutoAttendant.

Step 4 In the Extension field, enter the same extension that you provided for the CTI Route Point.

Step 5 Click Save.
The Edit Call Handler Basics (AutoAttendant) window is displayed.

Step 6 Edit the required fields and click Save.

Configure Caller Input Option

Procedure

Step 1 From Cisco Unity Connection Administration, from the Cisco Unity Connection tree on the left, navigate to Call Management and choose System Call Handlers.

Step 2 Click AutoAttendant.
The Edit Call Handler Basics (AutoAttendant) window is displayed.

Step 3 Choose Edit > Caller Inputs.
The Caller Input window is displayed.

Step 4 In the Key column, click 0.
The Edit Caller Input (0) window is displayed.

Step 5 Click the Call Handler radio button, choose Operator from the drop-down list, and click the Attempt Transfer radio button.

Step 6 Click Save.
The Updated Caller Input status message is displayed.

Step 7 Choose Edit > Caller Inputs.
The Caller Input window is displayed.

Step 8 In the Key column, click 1.
The Edit Caller Input (0) window is displayed.

Step 9 In the Conversation radio button, choose Caller System Transfer from the drop-down list.

Step 10 Click Save.
The **Updated Caller Input** status message is displayed.

---

## Configure Extension for Operator Call Handler

### Procedure

**Step 1** From Cisco Unity Connection Administration, from the Cisco Unity Connection tree on the left, navigate to **Call Management** and choose **System Call Handlers**.

**Step 2** Click **Operator**.

The Edit Call Handler Basics (Operator) window is displayed.

**Step 3** Enter the extension of the operator in the **Extension** field and click **Save**.

The **Updated Caller Input** status message is displayed.

---

## Modify Standard Call Transfer Rule for Operator

### Procedure

**Step 1** From Cisco Unity Connection Administration, from the Cisco Unity Connection tree on the left, navigate to **Call Management** and choose **System Call Handlers**.

**Step 2** Click **Operator**.

The Edit Call Handler Basics (Operator) window is displayed.

**Step 3** From the **Edit** menu, choose **Transfer Rules**.

The Transfer Rules window is displayed.

**Step 4** Click **Standard**.

The Edit Transfer Rule (Standard) window is displayed.

**Step 5** In the **Transfer Calls to** option, click the **Extension** radio button and enter the configured operator extension number.

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

---

## Update Default System Transfer Restriction Table

### Procedure

**Step 1** From Cisco Unity Connection Administration, from the Cisco Unity Connection tree on the left, navigate to **System Settings** and choose **Restriction Tables**.

**Step 2** Click **Default System Transfer**.

The Edit Restriction Table Basics (Default System Transfer) window is displayed.

**Step 3** Uncheck the check box in the **Blocked** column for 6 in the **Order** column.
Step 4  Click Save.

Cisco Unity Connection Auto-Attendant Troubleshooting

For information about troubleshooting Auto-Attendant using Cisco Unity Connection, see the following:


Cisco Unified CCX Configuration

Auto-Attendant comes standard with the five-seat bundle of Cisco Unified Contact Center Express (Unified CCX).

Note

For information about the supported versions of Cisco Unified CCX with Unified Communications Manager, see Cisco Collaboration Systems Release Summary Matrix for IP Telephony.

For information about getting started with scripts, see the Cisco Unified Contact Center Express Getting Started with Scripts.

Cisco Unified CCX Prerequisites

- Install and configure Cisco Unified CCX before you can use Auto-Attendant. Cisco Unified CCX controls the software and its connection to the telephony system.

- Configure users on Unified Communications Manager.

Cisco Unified CCX Auto-Attendant Task Flow

Auto-Attendant configuration tasks are completed in Cisco Unified Contact Center Express (Unified CCX). To view detailed steps for the following tasks, see Cisco Unified CCX Administration Guide and the Cisco Unified Contact Center Express Getting Started with Scripts respectively.

Before you begin

- Learn more about the Auto-Attendant feature by reviewing Auto-Attendant Overview, on page 133.
• Learn more about Cisco UCCX with Auto-Attendant functionality by reviewing Cisco Unified CCX Configuration, on page 138

• Review Cisco Unified CCX Prerequisites, on page 138.

<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 Unified CM Telephony call control groups.</td>
<td>The Unified CCX system uses Unified CM Telephony call control groups to pool together a series of CTI ports, which the system uses to serve calls as they arrive or depart from the Unified CCX server.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Add a Cisco Media Termination (CMT) dialog control group.</td>
<td>The Cisco Media subsystem is a subsystem of the Unified CCX Engine. The Cisco Media subsystem manages the CMT media resource. CMT channels are required for Unified CCX to be able to play or record media. The Cisco Media subsystem uses dialog groups to organize and share resources among applications. A dialog group is a pool of dialog channels in which each channel is used to perform dialog interactions with a caller, during which the caller responds to automated prompts by pressing buttons on a touch-tone phone.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure a Cisco script application.</td>
<td>The Unified CCX script applications are applications that are based on scripts created in the Unified CCX Editor. These applications come with every Unified CCX system and executes scripts that are created in the Unified CCX Editor.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Provision a Unified CM Telephony trigger.</td>
<td>A Unified CM Telephony trigger responds to calls that arrive on a specific route point by selecting telephony and media resources to serve the call and invoking an application script to handle the call.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Customize Auto-Attendant.</td>
<td>The Cisco Unified CCX Administration page allows you to modify any existing Auto-Attendant instance as necessary.</td>
</tr>
<tr>
<td></td>
<td>• Modify an existing Auto-Attendant instance</td>
<td></td>
</tr>
</tbody>
</table>

Caution: All media termination strings begin with "auto" and contain the same ID as the call control group—not the CMT dialog group. Perform this procedure if the default media termination is configured and the ID differs.
Cisco Unified CCX Auto-Attendant Troubleshooting

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Configure the Auto-Attendant prompts</td>
<td>Cisco Unified CCX allows you to customize the Auto-Attendant prompts from the Cisco Unified CCX Administration Media Configuration window. It allows you to record the welcome prompt, configure the welcome prompt, and upload a spoken name.</td>
</tr>
</tbody>
</table>

What to do next

If you have any problems configuring Cisco Unified CCX, see Cisco Unified CCX Auto-Attendant Troubleshooting, on page 140.

Cisco Unified CCX Auto-Attendant Troubleshooting

For information about troubleshooting Auto-Attendant using Cisco Unified CCX, see http://docwiki.cisco.com/wiki/List_of_Troubleshooting_Tips_for_Unified_CCX_7.0#Cisco_Unified_Communications_Manager_Automated_Attendant (applicable only for Windows version).

Cisco Unity Express Configuration

For information about Auto-Attendant Configuration using Cisco Unity Express, see the “Configuring Auto Attendants” chapter in Cisco Unity Express VoiceMail and Auto Attendant CLI Administrator Guide for 3.0 and Later Versions.

For information about deploying a sample Auto-Attendant script, see “Deployment of sample script aa.aef” chapter in the Getting Started with Cisco Unified IP IVR.

For information about an Auto-Attendant example, see “Auto Attendant Script Example” chapter in the Cisco Unity Express Guide to Writing and Editing Scripts for 7.0 and Later Versions.

For information about Auto-Attendant design considerations, see “Auto Attendant Design Considerations” chapter in the Cisco Unity Express Design Guide.

Cisco Unity Express Auto-Attendant Troubleshooting

For information about Auto-Attendant troubleshooting using Cisco Unity Connection, see the “Troubleshooting Cisco Unity Express Automated Attendant” in Excerpts from Cisco IP Communications Express: CallManager Express with Cisco Unity Express.
Manager Assistant

Cisco Unified Communications Manager Assistant Overview

The Unified Communications Manager Assistant feature is a plug-in that an assistant can use to handle calls on behalf of a manager, intercept manager calls, and route them appropriately.

Manager Assistant supports up to 3500 managers and 3500 assistants. To accommodate this number of users, you can configure up to three Manager Assistant applications in one Unified Communications Manager cluster and assign managers and assistants to each instance of the application.

Manager Assistant supports shared line support and proxy line support.

Manager Assistant Architecture

The Manager Assistant architecture comprises the following:

- **Cisco IP Manager Assistant service**—After you install Unified Communications Manager, activate this service from the Cisco Unified Serviceability interface.

- **Assistant Console interface**—Allows assistants to access the Manager Assistant features on their computer to handle calls for managers. The Manager Assistant handles calls for an assistant and for as many as 33 managers.

- **Cisco Unified IP Phone interface**: Managers and assistants use softkeys and the Cisco Unified IP Phone Services button to access the Manager Assistant features.

For more information, see chapter Manager Assistant, in Feature Configuration Guide for Cisco Unified Communications Manager.
**Manager Assistant Database Access Architecture**

The database stores all Manager Assistant configuration information. When the manager or assistant logs in, the Cisco IP Manager Assistant service retrieves all data that is related to the manager or assistant from the database and stores it in memory. The database includes two interfaces:

- **Manager interface**—The manager phone makes available the manager features except Manager Configuration. Manager Assistant automatically logs in a manager into the Cisco IP Manager Assistant service when the Cisco IP Manager Assistant service starts.

  **Note** Managers also have access to Unified Communications Manager features such as Do Not Disturb and Immediate Divert.

- **Assistant interface**—The assistant accesses the Manager Assistant features by using the Assistant Console application and the Cisco Unified IP Phone. The Assistant Console, an application, provides call-control functions such as answer, divert, transfer, and hold. The assistant uses the Assistant Console to log in and log out, to set up assistant preferences, and to display the Manager Configuration window that is used to configure manager preferences.

For more information, see chapter Manager Assistant, in Feature Configuration Guide for Cisco Unified Communications Manager.

**Softkeys**

Manager Assistant supports the following softkeys:

- Redirect
- Transfer to VoiceMail
- Do Not Disturb

Manager Assistant supports the following softkey templates:

- Standard Manager—Supports manager for proxy mode
- Standard Shared Mode Manager—Supports manager for shared mode
- Standard Assistant—Supports assistant in proxy or shared mode
- Standard User—The system makes call-processing (such as Hold and Dial) softkeys available with the Standard User template.

**Manager Assistant Shared Line Overview**

When you configure Manager Assistant in shared line mode, the manager and assistant share a directory number, for example, 8001. The assistant handles calls for a manager on the shared directory number. When a manager receives a call on 8001, both the manager phone and the assistant phone ring.

The Manager Assistant features that do not apply to shared line mode include Default Assistant Selection, Assistant Watch, Call Filtering, and Divert All Calls. An assistant cannot see or access these features on the Assistant Console application.
Manager Assistant Proxy Line Overview

When you configure Manager Assistant in proxy line mode, the assistant handles calls for a manager using a proxy number. The proxy number is not the directory number for the manager, but is an alternate number chosen by the system that an assistant uses to handle manager calls. In proxy line mode, a manager and an assistant have access to all features that are available in Manager Assistant, which include Default Assistant Selection, Assistant Watch, Call Filtering, and Divert All Calls.

Manager Assistant Prerequisites

- Manager Assistant supports the following browsers and platform:
  - Unified Communications Manager Assistant Administration and the Assistant Console are supported on Microsoft Internet Explorer 7.0 or later, Firefox 3.x or later, and Safari 4.x or later.
  - On a computer running Windows XP, Windows Vista, Windows 7, Windows 10, or Apple MAC OS X, customers can open one of the browsers specified above.

- To display Manager Assistant features in other languages, install the locale installer before you configure the Manager Assistant.

- The Assistant Console application is supported on computers that run Windows 7, Windows XP, or Windows Vista. It requires that the JRE 1.4.2_05 exist in Program Files\Cisco\Cisco Unified Communications Manager.

- You must configure the phones and users, and associated the devices to the users. In addition, for shared line appearances between managers and assistants, you must configure the same directory number on the manager primary line and assistant secondary line.

- To add managers and assistants in bulk, install the Unified Communications Manager Bulk Administration Tool. For more information, see the Bulk Administration Guide for Cisco Unified Communications Manager.

Manager Assistant Task Flow for Proxy Lines

Before you begin

- Review Manager Assistant Prerequisites, on page 143.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Run the Cisco Unified CM Assistant Configuration Wizard, on page 144</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Manager And Assign Assistant For Proxy Line, on page 151</td>
</tr>
</tbody>
</table>
Run the Cisco Unified CM Assistant Configuration Wizard

You can run the Cisco Unified CM Assistant Configuration Wizard to automatically create partitions, calling search spaces, and route points. The wizard also creates Bulk Administration Tool (BAT) templates for the manager phones, the assistant phones, and all other user phones. You can use the BAT templates to configure the managers, assistants, and all other users. For more information about BAT, see Bulk Administration Guide for Cisco Unified Communications Manager.

Before you begin

Ensure that the configuration wizard runs on the same server (the Unified Communications Manager server) as the Bulk Administration Tool.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose Application > Cisco Unified CM Assistant Configuration Wizard.

**Step 2**
Click Next to begin the Cisco Unified CM Assistant Configuration wizard process.

**Step 3**
In the Partition for Managers window, enter a name, provide a description, and then click Next. Alternatively, you can accept the default partition name and description.

**Step 4**
In the Partition for CTI Route Point window, enter a name, provide a description, and then click Next. Alternatively, you can accept the default CTI route point name.

**Step 5**
In the Partition for All Users window, enter a name, provide a description and then click Next. Alternatively, you can accept the default partition name and description for all users.

**Step 6**
In the Intercom Partition window, enter a name, provide a description, and then click Next. Alternatively, you can accept the default intercom partition name.

**Step 7**
In the Assistant Calling Search Space window, enter a name, and provide a description. Alternatively, you can use the default calling search space name and description. The Available Partitions and Selected Partitions boxes under the Route Partitions for this Calling Search Space automatically list Partitions for the Assistant Calling Search Space. You can accept the default values or you can choose the applicable partition from the Available Partitions box. Use the up and down arrows to move partitions from one box to the other.

**Step 8**
Click Next.
Step 9 In the **Everyone Calling Search Space** window, enter a name, and provide a description. Alternatively, you can accept the default calling search space name and description for everyone. The Available Partitions and Selected Partitions boxes under the Route Partitions for this Calling Search Space automatically list Partitions for the Assistant Calling Search Space. You can accept the default values or you can choose the applicable partition from the Available Partitions box. Use the up and down arrows to move partitions from one box to the other.

Step 10 Click **Next**.
If you have existing calling search spaces that are configured on the system, the **Existing Calling Search Spaces** window is displayed; otherwise, proceed to the next step.
Manager Assistant requires that the existing calling search spaces add the prefix **Generated Route Point** and **Generated Everyone** partitions. The Available Calling Search Spaces and Selected Calling Search Spaces boxes automatically list these partitions. Use the up and down arrows to move partitions from one box to the other.

*Note* The prefix that is added to the existing calling search spaces may change if the administrator has changed the names of the partitions.

Step 11 Click **Next**.
Step 12 In the **CTI Route Point** window, enter a name in the CTI route point name field; otherwise, use the default CTI route point name.
Step 13 From the drop-down list, choose the appropriate device pool.
Step 14 Enter a route point directory number; otherwise, use the default route point directory number.
Step 15 From the drop-down list, choose the appropriate numbering plan and then click **Next**.
Step 16 In the **Phone Services** window, enter the primary phone service name; otherwise, use the default Phone Service name.
Step 17 From the drop-down list, choose the primary Cisco Unified Communications Manager Assistant server or enter a server name or IP address.
Step 18 Enter the secondary phone service name; otherwise, use the default phone service name.
Step 19 From the drop-down list, choose the secondary Cisco Unified Communications Manager Assistant server or enter a server name or IP address and then click **Next**.
The **Confirmation** window is displayed. It provides all the information that you chose. If the information is not correct, you can cancel the configuration process or return to the previous configuration windows.

Step 20 Click **Finish**.
Upon completion, a final status window is displayed.
Any errors that the configuration wizard generates is sent to a trace file. Access this file by using the following CLI command: `file get activelog tomcat/logs/ccmadmin/log4j`

---

**What to do next**

The Cisco Unified CM Assistant Configuration Wizard only creates the Cisco IP Manager Assistant service parameters. You must enter the remaining service parameters manually. For service parameter information, see Manager Assistant Service Parameters for Proxy Line, on page 145.

**Manager Assistant Service Parameters for Proxy Line**

From Cisco Unified CM Administration, choose **System > Service Parameters**. Choose the server on which the Cisco IP Manager Assistant service is active and click **?** for detailed descriptions.
### Manager Assistant Service Parameters for Proxy Line

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco IP Manager Assistant (Active) Parameters</strong></td>
<td></td>
</tr>
<tr>
<td>CTIManager (Primary) IP Address</td>
<td>This parameter specifies the IP address of the primary CTIManager that this Cisco IPMA server uses to process calls. No default value.</td>
</tr>
<tr>
<td>CTIManager (Backup) IP Address</td>
<td>This parameter specifies the IP address of the backup CTIManager that this Cisco IPMA server uses to process calls when primary CTIManager is down. No default value.</td>
</tr>
<tr>
<td>Route Point Device Name for Proxy Mode</td>
<td>This parameter specifies the device name of the CTI route point that this Cisco IPMA server uses to intercept all calls to managers' primary lines for intelligent call routing. Cisco recommends that you use same CTI route point device for all servers running the IPMA service. You must configure the CTI route point device name if any manager or assistant will be configured to use proxy mode.</td>
</tr>
<tr>
<td>CAPF Profile Instance Id for Secure Connection to CTIManager</td>
<td>This service parameter specifies the Instance ID of the Application CAPF Profile for the application user IPMASecureSysUser that this Manager Assistant will use to open a secure connection to CTIManager. Configure this parameter if CTIManager Connection Security Flag is enabled.</td>
</tr>
<tr>
<td><strong>Clusterwide Parameters (Parameters that apply to all servers)</strong></td>
<td>Important Click Advanced to view the hidden parameters.</td>
</tr>
<tr>
<td>Cisco IPMA Server (Primary) IP Address</td>
<td>This parameter specifies the IP address of the primary Cisco IPMA server. No default value.</td>
</tr>
<tr>
<td>Cisco IPMA Server (Backup) IP Address</td>
<td>This parameter specifies the IP address of the backup Cisco IPMA server. The backup server provides IPMA service when the primary IPMA server fails. No default value.</td>
</tr>
<tr>
<td>Cisco IPMA Server Port</td>
<td>This parameter specifies the TCP/IP port on the Cisco IPMA servers to which the IPMA Assistant Consoles will open socket connections. You may change the parameter if a port conflict exists. Default value: 2912</td>
</tr>
<tr>
<td>Setting</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Cisco IPMA Assistant Console Heartbeat Interval | This parameter specifies the interval, in seconds, at which the Cisco IPMA server sends keepalive messages (commonly referred to as heartbeat) to the IPMA Assistant Consoles. The IPMA Assistant Consoles initiate failover when they fail to receive heartbeat from the server before the time that is specified in this parameter expires.  
Default value: 30 seconds |
| Cisco IPMA Assistant Console Request Timeout | This parameter specifies the time, in seconds, that the IPMA Assistant Consoles wait to receive a response from the Cisco IPMA server.  
Default value: 30 seconds |
| Cisco IPMA RNA Forward Calls                | This parameter determines whether Cisco IPMA Ring No Answer (RNA) forwarding is enabled. Valid values are True (Cisco IPMA forwards unanswered calls to next available assistant) or False (Cisco IPMA does not forward calls).  
This parameter works in conjunction with the Cisco IPMA RNA Timeout parameter; calls are forwarded after the time that is specified in the Cisco IPMA RNA Timeout parameter. If a voicemail profile is specified for the line, unanswered calls that cannot be forwarded to an assistant are sent to voicemail when this timer expires.  
Default value: False |
| Alpha Numeric UserID                         | This parameter determines whether Cisco IPMA Assistant Phone uses an alphanumeric user ID or a numeric user ID.  
Default value: True |
| Cisco IPMA RNA Timeout                       | This parameter specifies the time, in seconds, that the Cisco IPMA server waits before forwarding an unanswered call to the next available assistant. This parameter works in conjunction with the Cisco IPMA RNA Forward Calls parameter; forwarding occurs only if the Cisco IPMA RNA Forward Calls parameter is set to True.  
Default value: 10 seconds |
<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
</table>
| CTIManager Connection Security Flag | This parameter determines whether security for the Cisco IP Manager Assistant service CTIManager connection is enabled. If it is enabled, Cisco IPMA opens a secure connection to CTIManager using the CAPF profile that is configured for the instance ID (as specified in the **CAPF Profile Instance ID for Secure Connection to CTIManager** service parameter) for the application user IPMASecureSysUser.  
Default value: Non Secure  
To enable security, you must select an instance ID in the **CAPF Profile Instance ID for Secure Connection to CTIManager** service parameter. |
| Redirect call to Manager upon failure to reach Assistant | This parameter determines whether the Cisco Unified IP Manager Assistant application redirects the call back to the intended manager if the call fails to reach the selected proxy assistant.  
Default value: False |

**Advanced Clusterwide parameters**

*Important*  Configure unique IP addresses for each pool so that the same Cisco IPMA server IP address does not appear in more than one pool.

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
</table>
| Enable Multiple Active Mode | This parameter determines whether multiple instances of the Cisco IP Manager Assistant service must be run for scalability. If it is enabled, Cisco IPMA can run on the other nodes as configured in the Pool 2 and Pool 3 parameters.  
To enable multiple active mode, you must enter the IP addresses of the nodes on which you want to run the additional instances of Cisco IPMA. Configure the Cisco IP Manager Assistant service parameters on those nodes.  
Default value: False |
| Pool 2: Cisco IPMA Server (Primary) IP Address | If multiple active mode is enabled, this parameter specifies the IP address of the primary Cisco IPMA server of the second instance of Cisco IPMA.  
Configure the Cisco IP Manager Assistant service parameters on this node. |
<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pool 2: Cisco IPMA Server (Backup) IP Address</td>
<td>If multiple active mode is enabled, this parameter specifies the IP address of the backup Cisco IPMA server of the second instance of Cisco IPMA. The backup server provides IPMA service when the primary IPMA server fails. Configure the Cisco IP Manager Assistant service parameters on this node.</td>
</tr>
<tr>
<td>Pool 3: Cisco IPMA Server (Primary) IP Address</td>
<td>If multiple active mode is enabled, this parameter specifies the IP address of the primary Cisco IPMA server of the third instance of Cisco IPMA. Configure the Cisco IP Manager Assistant service parameters on this node.</td>
</tr>
<tr>
<td>Pool 3: Cisco IPMA Server (Backup) IP Address</td>
<td>If multiple active mode is enabled, this parameter specifies the IP address of the primary Cisco IPMA server of the third instance of Cisco IPMA. The backup server provides IPMA service when the primary IPMA server fails. Configure the Cisco IP Manager Assistant service parameters on this node.</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (Softkey Templates)**

**Important** Configure these parameters if you want to use the Manager Assistant automatic configuration for managers and assistants.

| Assistant Softkey Template | This parameter specifies the assistant softkey template that is assigned to assistant devices during Automatic Configuration. The value that is specified in this parameter is used when the **Automatic Configuration** check box is checked on the Cisco IPMA Assistant Configuration page. |
| Manager Softkey Template for Proxy Mode | This parameter specifies the manager softkey template for proxy mode that is assigned to manager devices during Automatic Configuration. This parameter applies only for managers that use proxy mode. |

**Clusterwide Parameters (IPMA Device Configuration Defaults for Proxy Mode)**

<p>| Manager Partition | This parameter defines the partition that is assigned to manager lines that IPMA handles on manager devices during Automatic Configuration. Make sure the partition you want to use has already been added to Cisco Unified CM Administration. If the <strong>Cisco IPMA Configuration Wizard</strong> is run, it will populate this value. This parameter applies only for managers that use proxy mode. |</p>
<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>All User Partition</td>
<td>This parameter specifies the partition that is configured on all proxy lines and the intercom line on assistant devices, as well as the intercom line on manager devices, during Automatic Configuration. Make sure the partition you want to use has already been added to Cisco Unified CM Administration. If the Cisco IPMA Configuration Wizard is run, it will populate this value. This parameter applies only for managers or assistants that use proxy mode.</td>
</tr>
<tr>
<td>IPMA Calling Search Space</td>
<td>This parameter specifies the calling search space that is configured for manager lines on manager devices that IPMA handles and the intercom line, as well as the assistant intercom line on assistant devices during Automatic Configuration. Make sure the calling search space you want to use has already been added to Cisco Unified CM Administration. If Cisco IPMA Configuration Wizard is run, it will populate this value. This parameter applies only for managers or assistants that use proxy mode.</td>
</tr>
<tr>
<td>Manager Calling Search Space</td>
<td>This parameter defines the manager calling search space that is configured on proxy lines on assistant devices during Automatic Configuration. This calling search space must be a calling search space that already exists in the system. If Cisco IPMA Configuration Wizard is run, it will populate this value. This parameter applies only for assistants that use proxy mode.</td>
</tr>
<tr>
<td>Cisco IPMA Primary Phone Service</td>
<td>This parameter defines the IP phone service to which manager/assistant devices will be subscribed during Automatic Configuration. If Cisco IPMA Configuration Wizard is run, it will populate this value. This parameter applies only for managers or assistants that use proxy mode.</td>
</tr>
<tr>
<td>Cisco IPMA Secondary Phone Service</td>
<td>This parameter defines the secondary IP phone service to which manager or assistant devices will be subscribed during Automatic Configuration. If Cisco IPMA Configuration Wizard is run, it will populate this value. This parameter applies only for managers or assistants that use proxy mode.</td>
</tr>
</tbody>
</table>

**Clusterwide Parameters (Proxy Directory Number Range for Proxy Mode)**
### Setting

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Starting Directory Number</strong></td>
</tr>
<tr>
<td>This parameter specifies the starting directory number that is used as the</td>
</tr>
<tr>
<td>starting number for automatic generation of proxy directory numbers during</td>
</tr>
<tr>
<td>IPMA assistant configuration. After an auto-generated proxy line number is</td>
</tr>
<tr>
<td>used for an assistant, the next number will be generated for the next</td>
</tr>
<tr>
<td>assistant, and so on. This parameter applies only for assistants that use</td>
</tr>
<tr>
<td>proxy mode.</td>
</tr>
</tbody>
</table>

| **Ending Directory Number**                                                   |
| This parameter specifies the ending directory number for automatic generation |
| of proxy directory numbers during IPMA assistant configuration. Configuration  |
| will stop at this number. This parameter applies only for assistants that use |
| proxy mode.                                                                   |

### Clusterwide Parameters (Proxy Directory Number Range for Proxy Mode)

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Number of Characters to be Stripped from Manager DN</strong></td>
</tr>
<tr>
<td>This parameter specifies the number of characters to be stripped from the</td>
</tr>
<tr>
<td>manager directory number (DN) in the process of generating the proxy DN.</td>
</tr>
<tr>
<td>Generating a proxy DN may involve stripping some number of digits and adding</td>
</tr>
<tr>
<td>a prefix. Digits are stripped starting from the left. This parameter applies</td>
</tr>
<tr>
<td>only for assistants that use proxy mode.</td>
</tr>
</tbody>
</table>

| **Prefix for Manager DN**                                                     |
| This parameter specifies the prefix to be added to a manager DN in the       |
| process of generating the proxy DN. Generating a proxy DN may involve some   |
| stripping of digits and adding a prefix. This parameter applies only for     |
| assistants that use proxy mode.                                              |

### Configure Manager And Assign Assistant For Proxy Line

For information about configuring a new user and associating a device to the user, see Administration Guide for Cisco Unified Communications Manager.

**Note**

Make sure you configure manager information before you configure assistant information for an assistant.

**Procedure**

- **Step 1** From Cisco Unified CM Administration, choose **User Management > End User**.
- **Step 2** Click **Find**.
  The search result displays all the end users that are configured in Unified Communications Manager.
- **Step 3** From the **Related Links** drop-down list, choose **Manager Configuration** and click **Go**.
Tip  To view existing assistant configuration information, click the assistant name in the Associated Assistants list and click View Details. The Cisco Unified CM Assistant - Assistant Configuration window is displayed. To return to the manager configuration information, click the manager name in the Associated Managers list and click View Details.

The Cisco Unified CM Assistant - Manager Configuration window is displayed.

Step 4  From the Device Name/Profile drop-down list, choose the device name or device profile to associate a device name or device profile with a manager. For more information about Extension Mobility with Manager Assistant, see Manager Assistant Interactions, on page 172.

Note  If the manager telecommutes, click the Mobile Manager check box and optionally choose a device profile from the Device Name/Profile drop-down list. After you choose a device profile, the manager must log in to the phone by using extension mobility before accessing Manager Assistant.

Step 5  From the Intercom Line drop-down list, choose the intercom line appearance for the manager, if applicable.

Note  The chosen intercom line applies to the Manager Assistant and Unified Communications Manager intercom features.

Step 6  From the Assistant Pool drop-down list, choose the appropriate pool number (1 to 3).

Step 7  From the Available Lines selection box, choose a line that you want Manager Assistant to control and click the down arrow to make the line display in the Selected Lines selection box. Configure up to five Manager Assistant—controlled lines.

Tip  To remove a line from the Selected Lines selection box and from Manager Assistant control, click the up arrow.

Step 8  Check the Automatic Configuration check box to automatically configure the softkey template, subscribe to the Manager Assistant phone service, calling search space, and partition for Manager Assistant—Controlled selected lines and intercom line; and Auto Answer with Speakerphone for intercom line for the manager phone based on the Cisco IP Manager Assistant service parameters.

Note  Automatic Configuration for intercom applies only when using the Manager Assistant intercom feature for the Cisco Unified IP Phones 7940 and 7960.

Step 9  Click Save.

If you checked the Automatic Configuration check box and the service parameters are invalid, a message displays. Ensure that the service parameters are valid. Upon successful completion of the automatic configuration, the manager device resets. If you configured a device profile, the manager must log out and log in to the device for settings to take effect.

Configure Assistant Line Appearances for Proxy Line

A proxy line specifies a phone line that appears on the assistant Cisco Unified IP Phone. Manager Assistant uses proxy lines to manage calls that are intended for a manager. The administrators can manually configure a line on the assistant phone to serve as the proxy line, or you can enable the Automatic Configuration check box to generate a DN and to add the line to the assistant phone.
1. Make sure that you configure manager information and assign an assistant to the manager before you configure assistant information for an assistant.

2. If you want to automatically configure proxy line on the assistant phone, configure the service parameters in **Proxy Directory Number Range** and **Proxy Directory Number Prefix** sections.

---

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **User Management > End User**.

**Step 2**
Click **Find**.

**Step 3**
Click on the user name to display user information for the chosen assistant.
The **End User Configuration** window is displayed.

**Step 4**
From the **Related Links** drop-down list, choose **Assistant Configuration** and click **Go**.

**Note**
The system automatically sets the softkey template and intercom line on the basis of the Cisco IP Manager Assistant service parameter settings when the **Automatic Configuration** check box is checked. In addition, the system also sets Auto Answer with Speakerphone for intercom line.

**Step 5**
From the **Device Name** drop-down list, choose the device name to associate with the assistant.

**Step 6**
From the **Intercom Line** drop-down list, choose the incoming intercom line appearance for the assistant.

**Step 7**
From the **Primary Line** drop-down list, choose the primary line for the assistant.

**Step 8**
To associate the manager line to the assistant line, perform the following steps from the Manager Association to Assistant Line selection box:

a) From the **Available Lines** drop-down list, choose the assistant line that will be associated with the manager line.

b) From the **Manager Names** drop-down list, choose the preconfigured manager name for whom this proxy line will apply.

c) From the **Manager Lines** drop-down list, choose the manager line for which this proxy line will apply.

**Step 9**
Click **Save**.
The update takes effect immediately. If you chose **Automatic Configuration**, the assistant device automatically resets.

---

**Manager Assistant Task Flow for Shared Lines**

**Before you begin**

- Review **Manager Assistant Prerequisites, on page 143**.
# Manager Assistant Task Flow for Shared Lines

## Procedure

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<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
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<td>Configure Partitions for Manager Assistant Shared Line Support, on page 155</td>
<td>Configure a partition for lines that is used by Manager Assistant.</td>
</tr>
<tr>
<td>2</td>
<td>Configure Calling Search Spaces for Manager Assistant Shared Line Support, on page 156</td>
<td>Configure calling search spaces for manager and assistant lines.</td>
</tr>
<tr>
<td>3</td>
<td>Configure Cisco IP Manager Assistant Service Parameters, on page 157</td>
<td>Configure these parameters to use automatic configuration for managers and assistants.</td>
</tr>
<tr>
<td>4</td>
<td>Configure Intercom Settings</td>
<td>Configure multiple pools if you need to support a large number of managers and assistants. You can configure up to three active Cisco IP Manager Assistant servers, with each managing up to 2500 pairs of managers and assistants.</td>
</tr>
<tr>
<td></td>
<td>- Configure an Intercom Partition, on page 158</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Configure an Intercom Calling Search Space, on page 303</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Configure an Intercom Directory Number, on page 304</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Configure an Intercom Translation Pattern, on page 304</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>Configure Multiple Manager Assistant Pool, on page 160</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>Configure Secure TLS Connection to CTI for Manager Assistant</td>
<td>Follow these procedures if your system is running in mixed mode.</td>
</tr>
<tr>
<td></td>
<td>- Configure IPMASecureSysUser Application User, on page 161</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Configure CAPF Profile, on page 161</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Configure Cisco WebDialer Web Service , on page 163</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>Configure CTI Route Point, on page 164</td>
<td>Cisco Unified Communications Manager Assistant requires creation of CTI route point to intercept and route calls from managers.</td>
</tr>
<tr>
<td>8</td>
<td>Configure IP Phone Services for Manager and Assistant, on page 164</td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>Configure Phone Button Templates for Manager, Assistant, and Everyone, on page 168</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>Configure Manager and Assign Assistant for Shared Line Mode, on page 169</td>
<td></td>
</tr>
<tr>
<td>11</td>
<td>Configure Assistant Line Appearances for Shared Line, on page 170</td>
<td></td>
</tr>
</tbody>
</table>
Configure Partitions for Manager Assistant Shared Line Support

You must create three partitions: Generated_Everyone, Generated_Managers, and Generated_Route_Point.

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 for Manager Assistant Shared Line Support

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 Calling Search Spaces for Manager Assistant Shared Line Support

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.

You must create two calling search spaces: Generated_CSS_I_E and Generated_CSS_M_E.

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.
Configure Cisco IP Manager Assistant Service Parameters

Configure Cisco IP Manager Assistant service parameters if you want to use the Manager Assistant automatic configuration for managers and assistants. You must specify the cluster-wide parameters once for all Cisco IP Manager Assistant services and general parameters for each Cisco IP Manager Assistant service that is installed.

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 IP Manager Assistant service is active.
Step 3 From the Service drop-down list, choose Cisco IP Manager Assistant service.
The Service Parameter Configuration window, which lists the parameters, is displayed.
Step 4 Configure the Cisco IP Manager Assistant Parameters, Clusterwide Parameters (Parameters that apply to all servers), and Clusterwide Parameters (Softkey Templates).
For detailed descriptions, see the online help.
Step 5 Click Save.

Configure Intercom Settings

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure an Intercom Partition, on page 158</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure an Intercom Calling Search Space, on page 158</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure an Intercom Directory Number, on page 159</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure an Intercom Translation Pattern, on page 159</td>
</tr>
</tbody>
</table>
Configure an Intercom Partition

Procedure

Step 1 In the Cisco Unified Communications Manager Administration window, choose Call Routing > Intercom > Intercom Route Partition.

The Find and List Intercom Partitions window appears.

Step 2 Click Add New.

An Add New Intercom Partition window appears.

Step 3 Under the Intercom Partition Information section, in the Name box, enter the name and description of the intercom partition that you want to add.

Note To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (,) to separate the partition name and description on each line. If a description is not entered, Unified Communications Manager uses the partition name as the description.

Step 4 Click Save.

Step 5 Locate the partition that you want to configure.

Intercom Partition Configuration window is displayed

Step 6 Configure the fields in the Intercom Partition Configuration field area. See the online help for more information about the fields and their configuration options.

Step 7 Click Save.

The Intercom Partition Configuration window appears.

Step 8 Enter the appropriate settings. For detailed information about the Intercom Partition Configuration parameters, see online help.

Step 9 Click Save.

Step 10 Click Apply Config.

Configure an Intercom Calling Search Space

Procedure

Step 1 In the menu bar, choose Call Routing > Intercom > Intercom Calling Search Space.

Step 2 Click the Add New button.

Step 3 Configure the fields in the Intercom Calling Search Space field area. See the online help for more information about the fields and their configuration options.

Step 4 Click Save.
Configure an Intercom Directory Number

Procedure

**Step 1** Choose Call Routing > Intercom > Intercom Directory Number.
The Find and List Intercom Directory Numbers window is displayed.

**Step 2** To locate a specific intercom directory number, enter search criteria and click Find.
A list of intercom directory numbers that match the search criteria displayed.

**Step 3** Perform one of the followings tasks:
  a) To add an intercom directory number, click Add New.
  b) To update an intercom directory number, click the intercom directory number to update.

The Intercom Directory Number Configuration window displayed.

**Step 4** Configure the fields in the Intercom Directory Number Configuration field area. For more information on the fields and their configuration options, see the system Online Help.

**Step 5** Click Save.

**Step 6** Click Apply Config.

**Step 7** Click Reset Phone.

**Step 8** Restart devices.

During the restart, the system may drop calls on gateways.

Configure an Intercom Translation Pattern

Procedure

**Step 1** Choose Call Routing > Intercom > Intercom Translation Pattern.
The Find and List Intercom Translation Patterns window appears.

**Step 2** Perform one of the followings tasks:
  a) To copy an existing intercom translation pattern, locate the partition to configure, click Copy eside the intercom translation pattern to copy.
  b) To add a new intercom translation pattern, click the Add New button.

**Step 3** Configure the fields in the Intercom Translation Pattern Configuration field area. For more information on the fields and their configuration options, see the system Online Help.

**Step 4** Click Save.

Ensure that the intercom translation pattern that uses the selected partition, route filter, and numbering plan combination 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.
Configure Multiple Manager Assistant Pool

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 IP Manager Assistant service is active.
Step 3  From the Service drop-down list, choose the Cisco IP Manager Assistant service.
Step 4  Click Advanced.
Step 5  Configure the following parameters to add multiple manager assistant pools in Clusterwide Parameters (Parameters that apply to all servers):
   a) Enable Multiple Active Mode—The default is False. When this parameter is set to True, the administrator can configure up to 7000 managers and assistants by using multiple pools.
   b) Pool 2: Cisco IPMA Server (Primary) IP Address—No default. The administrator must manually enter this IP address. Administrator can assign up to 2500 managers and assistants to this address.
   c) Pool 2: Cisco IPMA Server (Backup) IP Address—No default. The administrator must manually enter this IP address.
   d) Pool 3: Cisco IPMA Server (Primary) IP Address—No default. The administrator must manually enter this IP address and can assign up to 2500 managers and assistants to this address.
   e) Pool 3: Cisco IPMA Server (Backup) IP Address—No default. The administrator must manually enter this IP address.

For detailed descriptions, see the online help.

Step 6  Click Save.

Configure Secure TLS Connection to CTI for Manager Assistant

Manager Assistant uses WDSecureSysUser application user credentials to establish a secure TLS connection to CTI to make calls.

To configure the WDSecureSysUser application user to establish a secure TLS connection, complete the following tasks.

What to do next

Refer to the Manager Assistant Task Flow for Shared Lines, on page 153 to determine the next task to complete.
Before you begin

- Install and configure the Cisco CTL Client.
  For more information about CTL Client, see Security Guide for Cisco Unified Communications Manager.
- Verify that the Cluster Security Mode in the Enterprise Parameters Configuration window is 1 (mixed mode). Operating the system in mixed mode impacts other security functions in your system. If your system is not currently running in mixed mode, do not switch to mixed mode until you understand these interactions. For more information, see Security Guide for Cisco Unified Communications Manager.
- Verify that the Cluster SIPOAuth Mode field in the Enterprise Parameters Configuration window is set to Enabled.
- Activate the Cisco Certificate Authority Proxy Function (CAPF) service on the first node.

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 IPMASecureSysUser Application User, on page 161</td>
<td>Configure IPMASecureSysUser Application User.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure CAPF Profile, on page 161</td>
<td>Configure Certificate Authority Proxy Function (CAPF) Profile for the IPMASecureSysUser Application User.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Cisco WebDialer Web Service, on page 163</td>
<td>Configure service parameters for the Cisco IP Manager Assistant service.</td>
</tr>
</tbody>
</table>

Configure IPMASecureSysUser Application User

Use this procedure to configure IPMASecureSysUser application user.

Procedure

- Step 1 From Cisco Unified CM Administration, choose User Management > Application User.
- Step 2 Click Find.
- Step 3 From the Find and List Application Users Application window, choose WDSecureSysUser.
- Step 4 Configure the fields in the Application User Configuration window and click Save.

Configure CAPF Profile

Certificate Authority Proxy Function (CAPF) is a component that performs tasks to issue and authenticate security certificates. When you create an application user CAPF profile, the profile uses the configuration details to open secure connections for the application.
Procedure

Step 1  From Cisco Unified CM Administration, choose User Management > Application User CAPF Profile.

Step 2  Perform one of the following tasks:

- Click Add New in the Find window, to add a new CAPF profile.
- Click Copy for that record in the Copy column, to copy an existing profile, and locate the appropriate profile.

To update an existing entry, locate and display the appropriate profile.

Step 3  Configure or update the relevant CAPF profile fields. See the Related Topics section information about the fields and their configuration options.

Step 4  Click Save.

Step 5  Repeat the procedure for each application and end user that you want to use security.

Related Topics

CAPF Profile Settings, on page 162

CAPF Profile Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Application User</strong></td>
<td>From the drop-down list, choose the application user for the CAPF operation. This setting displays configured application users. This setting does not appear in the End User CAPF Profile window.</td>
</tr>
<tr>
<td><strong>End User ID</strong></td>
<td>From the drop-down list, choose the end user for the CAPF operation. This setting displays configured end users. This setting does not appear in the Application User CAPF Profile window.</td>
</tr>
<tr>
<td><strong>Instance ID</strong></td>
<td>Enter 1 to 128 alphanumeric characters (a-z, A-Z, 0-9). The Instance ID identifies the user for the certificate operation. You can configure multiple connections (instances) of an application. To secure the connection between the application and CTIManager, ensure that each instance that runs on the application PC (for end users) or server (for application users) has a unique certificate. This field relates to the CAPF Profile Instance ID for Secure Connection to CTIManager service parameter that supports web services and applications.</td>
</tr>
</tbody>
</table>
| **Certificate Operation** | From the drop-down list, choose one of the following options:  
  - **No Pending Operation**—This message is displayed when no certificate operation is occurring. (default setting)  
  - **Install/Upgrade**—This option installs a new certificate or upgrades an existing locally significant certificate for the application. |
Setting | Description
---|---
Authentication Mode | The authentication mode for the Install/Upgrade certificate operation specifies By Authentication String, which means CAPF installs, upgrades, or troubleshoots a locally significant certificate only when the user or administrator enters the CAPF authentication string in the JTAPI/TSP Preferences window.

Authentication String | To create your own authentication string, enter a unique string.
Each string must contain 4 to 10 digits.
To install or upgrade a locally significant certificate, the administrator must enter the authentication string in the JTAPI/TSP preferences GUI on the application PC. This string supports one-time use only; after you use the string for the instance, you cannot use it again.

Generate String | To automatically generate an authentication string, click this button. The 4-to-10-digit authentication string appears in the Authentication String field.

Key Size (bits) | From the drop-down list, choose the key size for the certificate. The default setting is 1024. The other option for key size is 512.
Key generation, which is set at low priority, allows the application to function while the action occurs. Key generation may take up to 30 or more minutes.

Operation Completes by | This field, which supports all certificate operations, specifies the date and time by which you must complete the operation.
The values that are displayed apply for the first node.
Use this setting with the CAPF Operation Expires in (days) enterprise parameter, which specifies the default number of days in which the certificate operation must be completed. You can update this parameter at any time.

Certificate Operation Status | This field displays the progress of the certificate operation, such as pending, failed, or successful.
You cannot change the information that is displayed in this field.

**Configure Cisco WebDialer Web Service**

**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 WebDialer Web service is active.

**Step 3**
From the Service drop-down list, choose the Cisco WebDialer Web service.
A list of parameters appears.

**Step 4**
Navigate to and update the CTIManager Connection Security Flag and CAPF Profile Instance ID for Secure Connection to CTIManager parameters.
To view parameter descriptions, click the parameter name link.

**Note**
CTIManager supports IPv4 and IPv6 addresses.
**Configure CTI Route Point**

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > CTI Route Point**.
2. Click **Add New**. The **CTI Route Point Configuration** window is displayed.
3. In the **Device Name** field, enter the device name.
4. From the **Device Pool** drop-down list, choose **Default**.
5. From the **Calling Search Space** drop-down list, choose **Generated_CSS_M_E**.
6. Check the **Use Device Pool Calling Party Transformation CSS** check box.
7. Click **Save**. The **Add successful** status message is displayed.
8. From the Association area, click **Line [1] - Add a new DN**. The **Directory Number Configuration** window is displayed.
9. Enter a directory number in the **Directory Number** field.
10. From the **Route Partition** drop-down list, choose **Generated_Route_Point**.
11. Click **Save**.

**Configure IP Phone Services for Manager and Assistant**

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > Device Settings > Phone Services**.
2. Click **Add New**. The **IP Phone Services Configuration** window is displayed.
3. For each supported phone for managers and assistants, enter the required fields and click **Save**. See the Related Topics section for more information about the fields and their configuration options. The **Update successful message** is displayed.

**Related Topics**

- [Cisco IP Phone Services Configuration Fields](#), on page 165
## Cisco IP Phone Services Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Service Information</strong></td>
<td></td>
</tr>
<tr>
<td>Service Name</td>
<td>Enter the name of the service. If the service is not marked as an enterprise subscription, the service name will display in areas where you can subscribe to a service, for example, under Cisco Unified Communications Self Care Portal. Enter up to 128 characters for the service name. For Java MIDlet services, the service name must exactly match the name that is defined in the Java Application Descriptor (JAD) file.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Unified Communications Manager allows you to create two or more IP phone services with identical names. Cisco recommends that you do not do so unless most or all phone users are advanced, or unless an administrator always configures the IP phone services. Be aware that if AXL or any third-party tool accesses the list of IP phone services for configuration, you must use unique names for IP phone services.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>When the service URL points to an external customized URL, you cannot localize the service name according to the device locale of the phone. The service name gets displayed in English alphabets only.</td>
</tr>
<tr>
<td>ASCII Service Name</td>
<td>Enter the name of the service to display if the phone cannot display Unicode.</td>
</tr>
<tr>
<td>Service Description</td>
<td>Enter a description of the content that the service provides. The description can include up to 50 characters in any language, but it cannot include double quotation marks (&quot;) or single quotation marks (').</td>
</tr>
</tbody>
</table>
### Field Description

**Service URL**
Enter the URL of the server where the IP phone services application is located. Make sure that this server remains independent of the servers in your Unified Communications Manager cluster. Do not specify a Unified Communications Manager server or any server that is associated with Unified Communications Manager (such as a TFTP server or directory database publisher server).

For the services to be available, the phones in the Unified Communications Manager cluster must have network connectivity to the server.

For Cisco-signed Java MIDlets, enter the location where the JAD file can be downloaded; for example, a web server or the back-end application server to which the Java MIDlet communicates.

For Cisco-provided default services, the service URL is displayed as `Application:Cisco/<name of service>` by default; for example, `Application:Cisco/CorporateDirectory`. If you modify the service URL for Cisco-provided default services, verify that you configured both for the Service Provisioning setting, which displays in the **Phone**, **Enterprise Parameter**, and **Common Phone Profile Configuration** windows. For example, you use a custom corporate directory, so you change `Application:Cisco/CorporateDirectory` to the external service URL for your custom directory; in this case, change the Service Provisioning value to **Both**.

**Secure-Service URL**
Enter the secure URL of the server where the Cisco Unified IP Phone services application is located. Make sure that this server remains independent of the servers in your Unified Communications Manager cluster. Do not specify a Unified Communications Manager server or any server that is associated with Unified Communications Manager (such as a TFTP server or publisher database server).

For the services to be available, the phones in the Unified Communications Manager cluster must have network connectivity to the server.

*Note* If you do not provide a Secure-Service URL, the device uses the nonsecure URL. If you provide both a secure URL and a nonsecure URL, the device chooses the appropriate URL, based on its capabilities.

**Service Category**
Choose a service application type (XML or Java MIDlet).

If you choose Java MIDlet, when the phone receives the updated configuration file, the phone retrieves the Cisco-signed MIDlet application (JAD and JAR) from the specified Service URL and installs the application.

**Service Type**
Choose whether the service is provisioned to the Services, Directories, or Messages button or option on the phone; that is, if the phone has these buttons or options. To determine whether your phone supports these buttons or options, see the *Cisco Unified IP Phone Administration Guide* that supports your phone model.
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Service Vendor</td>
<td>This field allows you to specify the vendor or manufacturer for the service. This field is optional for XML applications, but it is required for Cisco-signed Java MIDlets. For Cisco-signed Java MIDlets, the value that you enter in this field must exactly match the vendor that is defined in the MIDlet JAD file. This field displays as blank for Cisco-provided default services. You can enter up to 64 characters.</td>
</tr>
</tbody>
</table>
| Service Version    | Enter the version number for the application. For XML applications, this field is optional and is informational only. For Cisco-signed Java MIDlets, consider the following information:  
• If you enter a version, the service version must exactly match the version that is defined in the JAD file. If you enter a version, the phone attempts to upgrade or downgrade the MIDlet if the version is different than what is installed on the phone.  
• If the field is blank, the version gets retrieved from the Service URL. Leaving the field blank ensures that the phone attempts to download the JAD file every time that the phone reregisters to Unified Communications Manager as well as every time that the Cisco-signed Java MIDlet is launched; this ensures that the phone always runs the latest version of the Cisco-signed Java MIDlet without you having to manually update the Service Version field.  
This field displays as blank for Cisco-provided default services. You can enter numbers and periods in this field (up to 16 ASCII characters). |
<p>| Enable             | This check box allows you to enable or disable the service without removing the configuration from Cisco Unified CM Administration (and without removing the service from the database). Uncheck the check box to remove the service from the phone configuration file and the phone. |</p>
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters</td>
<td>This pane lists the service parameters that apply to this IP phone service. Use the following buttons to configure service parameters for this pane:</td>
</tr>
<tr>
<td></td>
<td>• <strong>New Parameter</strong>—Click this button to display the <strong>Configure Cisco Unified IP Phone Service Parameter</strong> window, where you configure a new service parameter for this IP phone service.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Edit Parameter</strong>—Highlight a service parameter that is displayed in the Parameters pane, then click this button to display the <strong>Configure Cisco Unified IP Phone Service Parameter</strong> window, where you can edit the selected service parameter for this IP phone service.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Delete Parameter</strong>—Highlight a service parameter that is displayed in the Parameters pane, then click this button to delete a service parameter for this IP phone service. A popup window asks you to confirm deletion.</td>
</tr>
</tbody>
</table>

## Configure Phone Button Templates for Manager, Assistant, and Everyone

The procedures in this section describe how to configure phone button for manager and assistant.

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Perform this step to assign manage and assistant button features to line or speed dial keys.</td>
</tr>
<tr>
<td>Configure a Phone Button Template for Manager Assistant, on page 168</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Perform this step to configure the manager and assistant button for a phone.</td>
</tr>
<tr>
<td>Associate a Manager Assistant Button Template with a Phone, on page 169</td>
<td></td>
</tr>
</tbody>
</table>

## Configure a Phone Button Template for Manager Assistant

Use this procedure to configure a phone button template for the Manager Assistant feature.

### 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.
From the Line drop-down list, choose feature that you want to add to the template.

Click Save.

Perform one of the following tasks:

- If you modified a template that is already associated with devices, click Apply Config to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

### Associate a Manager Assistant Button Template with a Phone

**Before you begin**

Configure a Phone Button Template for Manager Assistant, on page 168

**Procedure**

1. From Cisco Unified CM Administration, choose Device > Phone.
2. Click Find to display the list of configured phones.
3. Choose the phone to which you want to add the phone button template.
4. In the Phone Button Template drop-down list, choose the phone button template that contains the new feature button.
5. Click Save.
   A dialog box is displayed with a message to press Reset to update the phone settings.

### Configure Manager and Assign Assistant for Shared Line Mode

**Procedure**

1. From Cisco Unified CM Administration, choose User Management > End User.
2. Click Find.
   The search result displays all the end users that are configured in Unified Communications Manager.
3. From the Related Links drop-down list, choose Manager Configuration and click Go.
4. Check the Automatic Configuration check box to automatically configure the softkey template and Auto Answer with Speakerphone for intercom line for the manager phone based on the Cisco IP Manager Assistant service parameters.

   **Note** Automatic Configuration for intercom applies only when the Unified Communications Manager Assistant intercom feature is used for the Cisco Unified IP Phones 7940 and 7960.

5. Check Uses Shared Lines check box.
6. From the Device Name/Profile drop-down list, choose the device name or device profile to associate a device name or device profile with a manager.
If the manager telecommutes, check the **Mobile Manager** check box and optionally choose a device profile from the **Device Name/Profile** drop-down list. When device profile is chosen, the manager must log in to the phone by using Cisco Extension Mobility before accessing Manager Assistant.

See the related topics for more information about Extension Mobility with Manager Assistant.

**Step 7**
From the **Intercom Line** drop-down list, choose the intercom line appearance for the manager, if applicable. The chosen intercom line applies to the Manager Assistant and Unified Communications Manager intercom features.

**Step 8**
From the **Assistant Pool** drop-down list, choose the appropriate pool number (1 to 3).

**Step 9**
Choose the name of the assistant from the Available Assistants selection box and move it to the Associated Assistants selection box by clicking the down arrow to assign an assistant to the manager.

You can go to the **Assistant Configuration** window by highlighting the assistant name and clicking the **View Details** link.

**Step 10**
Choose the appropriate line from the Available Lines list box and move it to the Selected Lines list box by clicking the down arrow to configure the Manager Assistant controlled lines. Make sure that the controlled line is always the shared line DN.

**Step 11**
Click **Save**.
If you checked the **Automatic Configuration** check box and the service parameters are invalid, a message is displayed. Ensure that the service parameters are valid. After successful completion of the automatic configuration, the manager device resets. If you configured a device profile, the manager must log out and log in to the device for the changes to take effect.

---

**Configure Assistant Line Appearances for Shared Line**

Administrators can set up one or more lines with a shared line appearance. The Unified Communications Manager system considers a directory number to be a shared line if it appears on more than one device in the same partition.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **User Management > End User**.

**Step 2**
Click **Find**.
The search result displays all the end users that are configured in Unified Communications Manager.

**Step 3**
Click on the username to display user information for the chosen assistant. The **End User Configuration** window is displayed.

**Step 4**
From the **Related Links** drop-down list, choose **Assistant Configuration** and click **Go**.
The **Assistant Configuration** window is displayed. The system automatically sets the softkey template and intercom line on the basis of the Cisco IP Manager Assistant service parameter settings when you check the **Automatic Configuration** check box. In addition, the system also sets Auto Answer with Speakerphone for intercom line.

**Step 5**
From the **Device Name** drop-down list, choose the device name to associate with the assistant.

**Step 6**
From the **Intercom Line** drop-down list, choose the incoming intercom line appearance for the assistant.

**Step 7**
From the **Primary Line** drop-down list, choose the primary line for the assistant.
a) To view existing manager configuration information, highlight the manager name in the Associated Managers list and click View Details. The Manager Configuration window is displayed.

b) To return to the Assistant Configuration window, highlight the assistant name and click View Details link in the Manager Configuration window.

In the Associated Manager selection list box, the name of the previously configured manager is displayed.

**Step 8**
To associate the manager line to the assistant line, perform the following steps from the Manager Association to Assistant Line selection box:

a) From the Available Lines drop-down list, choose the assistant line that will be associated with the manager line.

b) From the Manager Names drop-down list, choose the preconfigured manager name for whom this proxy line will apply.

c) From the Manager Lines drop-down list, choose the manager line for which this proxy line will apply.

**Step 9**
Click Save. The update takes effect immediately. If you chose Automatic Configuration, the assistant device automatically resets.

---

### Install Assistant Console Plugin

#### Procedure

**Step 1**
From Cisco Unified CM Administration, choose Application > Plugins. The Find and List Plugins window is displayed.

**Step 2**
Click Find. A list of installable application plug-ins is displayed.

**Step 3**
Click on the Download link for Cisco Unified CM Assistant Console and save the executable to a location.

**Step 4**
Run the executable file.

**Note** If you install the application on a Windows Vista PC, a security window may be displayed. Allow the installation to continue.

The Cisco Unified CallManager Assistant Console installation wizard is displayed.

**Step 5**
In the Introduction window, click Next.

**Step 6**
In the License Agreement window, click Next.

**Step 7**
Choose a location where you want the application to install and click Next. By default, the application installs in C: \Program Files\Cisco\ Unified CallManager Assistant Console.

**Step 8**
In the Pre-installation Summary window, review the summary and click Install. The installation begins.

**Step 9**
After the installation is complete, click Finish.

**Step 10**
Provide the assistant the username and password that is required to log in to the console.

**Step 11**
To launch the Assistant Console, click the desktop icon or choose Cisco Unified Communications Manager Assistant > Assistant Console from the Start...Programs menu.
Step 12  The Advanced tab in the Cisco Unified Communications Manager Assistant Settings window allows you to enable trace for the Assistant Console.

Step 13  Provide the assistant with the port number and the IP address or hostname of the Unified Communications Manager server on which the Cisco IP Manager Assistant service is active. The first time that the assistant logs in to the console, the assistant must enter the information in the Cisco Unified Communications Manager Assistant Server Port and the Cisco Unified Communications Manager Assistant Server Hostname or IP Address fields.

Manager Assistant Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bulk Administration Tool</td>
<td>You can use the Bulk Administration Tool to add many users (managers and assistants) at once instead of adding users individually. The Bulk Administration Tool templates that the Cisco Unified CM Assistant Configuration Wizard creates for Cisco Unified IP Phones support only the Unified Communications Manager intercom lines. For more information, see the Bulk Administration Guide for Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Calling Party Normalization</td>
<td>Manager Assistant automatically supports localized and globalized calls if you configure the Calling Party Normalization feature. Manager Assistant can display localized calling party numbers on the user interfaces. In addition, for an incoming call to the manager, Manager Assistant can display localized and globalized calling party numbers when filter pattern matching occurs.</td>
</tr>
<tr>
<td>Extension Mobility</td>
<td>You can simultaneously use Manager Assistant with the Cisco Extension Mobility feature. When you log in to the Cisco Unified IP Phone using Extension Mobility, the Cisco IP Manager Assistant service is automatically enabled on that phone. You can then access the Manager Assistant features. For more information about Cisco Extension Mobility, see Extension Mobility Overview, on page 393.</td>
</tr>
<tr>
<td>Internet Protocol Version 6 (IPv6)</td>
<td>Manager Assistant does not support IPv6, so you cannot use phones with an IP Addressing Mode of IPv6 Only with Manager Assistant. To use Manager Assistant with the phone, ensure that you configure the phone with an IP Addressing Mode of IPv4 Only or IPv4 and IPv6.</td>
</tr>
</tbody>
</table>
Manager Assistant provides statistical information in the CDR Analysis and Reporting (CAR) tool and provides a summary of changes to configurations in a change log.

The administrator can view a summary of changes that are made to the Manager or Assistant Configurations in Unified CM AssistantChangeLog*.txt. A manager can change defaults by accessing the Manager Configuration from a URL. An assistant can change the manager defaults from the Assistant Console. For information about the URL and Manager Configuration, see the *Cisco Unified Communications Manager Assistant User Guide*.

When the manager or assistant makes changes, the changes are sent to a log file called `ipma_changeLogxxx.log`. The log file resides on the server that runs the Cisco IP Manager Assistant service. Use the following command to obtain the log file: `file get activelog tomcat/logs/ipma/log4j/`

For more information about downloading the log file, see the *Cisco Unified Real-Time Monitoring Tool Administration Guide*.

Manager Assistant supports call-completion statistics and inventory reporting for managers and assistants. The CAR tool supports call-completion statistics. Cisco Unified Serviceability supports inventory reporting.

For more information, see the following guides:

- *Cisco Unified Serviceability Administration Guide*
- *Call Reporting and Billing Administration Guide for Cisco Unified Communications Manager*

The following points describe the interactions between Manager Assistant with shared line support and MLPP:

- The system preserves call precedence in the handling of calls by Manager Assistant. For example, when an assistant diverts a call, the system preserves the precedence of the call.

- Filtering of precedence calls occurs in the same manner as all other calls. The precedence of a call will not affect whether a call is filtered.

- Because Manager Assistant does not have information about the precedence of a call, it does not provide any additional indication of the precedence of a call on the Assistant Console.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reporting tools</td>
<td>Manager Assistant provides statistical information in the CDR Analysis and Reporting (CAR) tool and provides a summary of changes to configurations in a change log. The administrator can view a summary of changes that are made to the Manager or Assistant Configurations in Unified CM AssistantChangeLog*.txt. A manager can change defaults by accessing the Manager Configuration from a URL. An assistant can change the manager defaults from the Assistant Console. For information about the URL and Manager Configuration, see the <em>Cisco Unified Communications Manager Assistant User Guide</em>. When the manager or assistant makes changes, the changes are sent to a log file called <code>ipma_changeLogxxx.log</code>. The log file resides on the server that runs the Cisco IP Manager Assistant service. Use the following command to obtain the log file: <code>file get activelog tomcat/logs/ipma/log4j/</code>. For more information about downloading the log file, see the <em>Cisco Unified Real-Time Monitoring Tool Administration Guide</em>.</td>
</tr>
<tr>
<td>CDR Analysis and Reporting</td>
<td>Manager Assistant supports call-completion statistics and inventory reporting for managers and assistants. The CAR tool supports call-completion statistics. Cisco Unified Serviceability supports inventory reporting. For more information, see the following guides: <em>Cisco Unified Serviceability Administration Guide</em>  <em>Call Reporting and Billing Administration Guide for Cisco Unified Communications Manager</em></td>
</tr>
<tr>
<td>Multilevel Precedence and Preemption (MLPP)</td>
<td>The following points describe the interactions between Manager Assistant with shared line support and MLPP:  - The system preserves call precedence in the handling of calls by Manager Assistant. For example, when an assistant diverts a call, the system preserves the precedence of the call.  - Filtering of precedence calls occurs in the same manner as all other calls. The precedence of a call will not affect whether a call is filtered.  - Because Manager Assistant does not have information about the precedence of a call, it does not provide any additional indication of the precedence of a call on the Assistant Console.</td>
</tr>
</tbody>
</table>
Manager Assistant Restrictions

### Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1

---

### Manager Assistant Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Assistant Console Application</td>
<td>To install the Assistant Console application on a computer with Microsoft Internet Explorer 7 (or later), install the Microsoft Java Virtual Machine (JVM) before the Assistant Console installation.</td>
</tr>
<tr>
<td>Call Management features</td>
<td>The Assistant Console does not support hunt groups or queues, recording and monitoring, one-touch Call Pickup, and On-Hook transfer (the ability to transfer a call by pressing the Transfer softkey and going on hook to complete the transfer).</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager Assistant Troubleshooting

This section describes the troubleshooting tools for Manager Assistant and the client desktop, and troubleshooting information for Manager Assistant.

---

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IP Phones</td>
<td>Manager Assistant supports SIP on Cisco Unified IP Phones 7900 Series, except for Cisco Unified IP Phones 7940 and 7960. Manager Assistant supports up to 3500 managers and 3500 assistants by configuring multiple Cisco IP Manager Assistant servers (pools). When you enable multiple pools, the manager and all configured assistants for that manager should belong to the same pool. Cisco Unified IP Phones 7960 and 7940 support only the Unified Communications Manager Assistant Intercom lines feature. Cisco Unified IP Phones 7900 (except 7940 and 7960) support only the Unified Communications Manager Intercom feature. One manager can have up to ten assigned assistants and one assistant can support up to 33 managers (if each manager has one Unified Communications Manager–controlled line). Only one assistant at a time can assist a manager. Manager Assistant supports up to 3500 managers and 3500 assistants per Unified Communications Manager cluster.</td>
</tr>
<tr>
<td>Intercom</td>
<td>After an upgrade, Manager Assistant users that use the incoming intercom line do not get upgraded automatically to the Unified Communications Manager Intercom feature. The system does not support calls between the Unified Communications Manager Intercom feature and regular lines (which may be configured as Manager Assistant Intercom lines).</td>
</tr>
<tr>
<td>Single Sign-On</td>
<td>Manager Assistant is not supported in the Single Sign-On environment.</td>
</tr>
<tr>
<td>Speed Dial</td>
<td>Cisco Unified IP Phones 7940, 7942, and 7945 support only two lines or speed-dial buttons.</td>
</tr>
</tbody>
</table>
### LocationTool Description

<table>
<thead>
<tr>
<th>Tool Description</th>
<th>Location</th>
</tr>
</thead>
</table>
| Cisco Unified CM Assistant server trace files | The trace files reside on the server that runs the Cisco IP Manager Assistant service. You can download these files from the server using one of the following methods:  
- Use the CLI command `file get activelog tomcat/logs/ipma/log4j`.  
- Use the trace collection features in the Cisco Unified Real-Time Monitoring Tool (RTMT). For more information, see the *Cisco Unified Real-Time Monitoring Tool Administration Guide*.  
You can enable debug tracing by choosing *Cisco Unified Serviceability > Trace > Configuration*. |
| Cisco IPMA client trace files    | $INSTALL_DIR/logs/ACLog*.txt on the client desktop, in the same location where the Unified CM Assistant assistant console resides. To enable debug tracing, go to the *Settings* dialog box in the Assistant Console. In the *Advanced* panel, check the *Enable Trace* check box.  
Note: This check box enables only debug tracing. Error tracing always remains On. |
| Cisco IPMA client install trace files | $INSTALL_DIR/InstallLog.txt on the client desktop, in the same location where the Assistant Console resides. |
| Cisco IPMA Client AutoUpdater trace files | $INSTALL_DIR/UpdatedLog.txt on the client desktop, in the same location where the Unified CM Assistant Console resides. |
| Install directory                | By default—C:\Program Files\Cisco\Unified Communications Manager Assistant Console\ |

---

### Calling Party Gets Reorder Tone

#### Problem

Calling party gets a reorder tone or a message:

*This call cannot be completed as dialed.*

#### Possible Cause

The calling search space of the calling line may not be configured correctly.

#### Solution

Check the calling search space of the line. For more information about configuration, see the *System Configuration Guide for Cisco Unified Communications Manager*.

You can also use the Cisco Dialed Number Analyzer service to check for flaws in the calling search space. For more information, see the *Cisco Unified Communications Manager Dialed Number Analyzer Guide*.  

---

*Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1*
Calls Do Not Get Routed When Filtering Is On or Off

Problem
Calls are not routed properly.

Possible Cause 1
Cisco CTI Manager service may have stopped.

Solution 1
Restart the Cisco CTI Manager and Cisco IP Manager Assistant services from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

Possible Cause 2
The Unified Communications Manager Assistant route point was not configured properly.

Solution 2
Use wildcards to match the directory number of the Unified Communications Manager Assistant CTI route point and the primary directory numbers of all managers that are configured for Unified Communications Manager Assistant.

Possible Cause 3
The status window on the manager phone displays the message Filtering Down. This message can indicate that Unified Communications Manager Assistant CTI route point may be deleted or may not be in service.

Solution 3
Use the following procedure to configure the CTI route point and restart the Cisco IP Manager Assistant service:

1. From Cisco Unified CM Administration, choose Device > CTI Route Point.
2. Find the route point, or add a new route point. For more information about configuration, see the System Configuration Guide for Cisco Unified Communications Manager.
3. Restart the Cisco CTI Manager and Cisco IP Manager Assistant services from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

Cisco IP Manager Assistant Service Unreachable

Problem
After you open the Assistant Console, the following message is displayed:

Cisco IPMA Service Unreachable
**Possible Cause 1**
Cisco IP Manager Assistant service may have stopped.

**Solution 1**
Restart the Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

**Possible Cause 2**
The server address for the primary and secondary Unified Communications Manager Assistant servers may be configured as DNS names, but the DNS names are not configured in the DNS server.

**Solution 2**
Use the following procedure to replace the DNS name.
1. From Cisco Unified CM Administration, choose System > Server.
2. Replace the DNS name of the server with the corresponding IP address.
3. Restart the Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

**Possible Cause 3**
The Cisco CTI Manager service may have stopped.

**Solution 3**
Restart the Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

**Possible Cause 4**
The Unified Communications Manager Assistant service might be configured to open a CTI connection in secure mode, but the security configuration may not be complete.

If this scenario occurs, the following message is displayed in the alarm viewer or in the Unified Communications Manager Assistant service logs:

**IPMA Service cannot initialize – Could not get Provider.**

**Solution 4**
Check the security configuration in the service parameters of Cisco IP Manager Assistant service.

Restart the Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.
Cannot Initialize Cisco IP Manager Assistant Service

Problem
The Cisco IP Manager Assistant service cannot open a connection to CTI Manager, and the following message is displayed:

IPMA Service cannot initialize - Could not get Provider

Possible Cause
The Cisco IP Manager Assistant service cannot open a connection to CTIManager. You can see the message in the alarm viewer or in the Unified CM Assistant service logs.

Solution
Restart the Cisco CTI Manager and Cisco IP Manager Assistant services from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

Assistant Console Installation from Web Fails

Problem
Assistant Console installation from the web fails. The following message is displayed:

Exception: java.lang.ClassNotFoundException: InstallerApplet.class

Possible Cause
Using the Sun Java plug-in virtual machine instead of the Microsoft JVM with the standard Unified Communications Manager Assistant Console install causes failures.

Solution
The administrator directs the user to the following URL, which is a JSP page that supports the Sun Java plug-in:

https://<servername>:8443/ma/Install/IPMAConsoleInstallJar.jsp

HTTP Status 503—This Application Is Not Currently Available

Problem
http://<server-name>:8443/ma/Install/IPMAConsoleInstall.jsp displays the following error message:

HTTP Status 503—This application is not currently available

Possible Cause
Cisco IP Manager Assistant service has not been activated or is not running.
Manager Is Logged Out While the Service Is Still Running

Problem
Although the manager is logged out of Unified Communications Manager Assistant, the service still runs. The display on the manager IP phone disappears. Calls do not get routed, although filtering is On. To verify that the manager is logged out, view the application log using the Cisco Unified Real-Time Monitoring Tool. Look for a warning from the Cisco Java Applications that indicates that the Cisco IP Manager Assistant service logged out.

Possible Cause
The manager pressed the softkeys more than four times per second (maximum limit allowed).

Solution
The Unified Communications Manager administrator must update the manager configuration. Perform the following procedure to correct the problem:

1. From Cisco Unified CM Administration, choose User Management > End User.
   The Find and List Users window is displayed.
2. Enter the manager name in the search field and click Find.
3. From the search results list, choose the manager that you want to update.
   The End User Configuration window is displayed.
4. From the Related Links drop-down list, choose Cisco IPMA Manager and click Go.
5. Make the necessary changes to the manager configuration and click Update.

Manager Cannot Intercept Calls That Are Ringing on the Assistant Proxy Line

Problem
The manager cannot intercept the calls that are ringing on the assistant proxy line.

Possible Cause
The calling search space of the proxy line is not configured properly.
Solution
Check the calling search space of the proxy line for the assistant phone. Perform the following procedure to correct the problem:

1. From Cisco Unified CM Administration, choose **Device > Phone**.
   The **Find and List Phones** search window is displayed.

2. Click the assistant phone.
   The **Phone Configuration** window is displayed.

3. Verify the calling search space configuration for the phone and for the directory number (line) and update as appropriate.

No Page Found Error

Problem
http://<server-name>:8443/ma/Install/IPMAConsoleInstall.jsp displays the following error message:
No Page Found Error

Possible Cause 1
Network problems.

Solution 1
Ensure that the client has connectivity to the server. Ping the server name that is specified in the URL and verify that it is reachable.

Possible Cause 2
Misspelled URL.

Solution 2
Because URLs are case sensitive, ensure that the URL matches exactly with the URL in the instructions.

System Error - Contact System Administrator

Problem
After you open the Assistant Console, the following message is displayed:
System Error - Contact System Administrator

Possible Cause 1
You may have upgraded the Unified Communications Manager. The system does not upgrade the Assistant Console automatically when you upgrade the Unified Communications Manager.
Solution 1
Uninstall the console by choosing Start > Programs > Cisco Unified Communications Manager Assistant > Uninstall Assistant Console and reinstall the console from URL https://<server-name>:8443/ma/Install/IPMAConsoleInstall.jsp.

Possible Cause 2
The user is not configured correctly in the database.

Solution 2
Ensure that the user ID and the password are run as a Unified Communications Manager user through Cisco Unified CM Administration.

Possible Cause 3
When you deleted a manager from an assistant, Cisco Unified CM Administration left a blank line for the assistant.

Solution 3
From the Assistant Configuration window, reassign the proxy lines.

Unable to Call Manager When Cisco IP Manager Assistant Service is Down

Problem
Calls do not get routed properly to managers when Cisco IP Manager Assistant service goes down.

Possible Cause
The Unified Communications Manager Assistant CTI route point does not have Call Forward No Answer enabled.

Solution
Perform the following procedure to properly configure the Unified Communications Manager Assistant route point.
1. From Cisco Unified CM Administration, choose Device > CTI Route Point.
   The Find and List CTI Route Point search window is displayed.
2. Click Find.
   A list of configured CTI route points is displayed.
3. Choose the Unified Communications Manager Assistant CTI route point that you want to update.
4. In the CTI Route Point Configuration window, click the line to update from the Association area.
5. In the Call Forward and Pickup Settings section, check the Forward No Answer Internal and the Forward No Answer External check box and enter the CTI route point DN in the Coverage/Destination field (for example, CFNA as 1xxx for the route point DN 1xxx).
6. In the Calling Search Space drop-down list, choose CSS-M-E (or appropriate calling search space).
7. Click Update.

User Authentication Fails

Problem
User authentication fails when you sign in using the login window from the Assistant Console.

Possible Cause
The following probable causes can apply:

• Incorrect management of the user in the database
• Incorrect management of the user as an assistant or a manager

Solution
Ensure that the user ID and the password are ran as a Unified Communications Manager user through Cisco Unified CM Administration.

You must run the user as an assistant or a manager by associating the Unified Communications Manager Assistant user information, which you access through Cisco Unified CM Administration under User Management > End User.
PART VI

Voice Messaging Features

- Audible Message Waiting Indicator, on page 187
- Immediate Divert, on page 191
Audible Message Waiting Indicator Overview

You can configure Audible Message Waiting Indicator (AMWI) to play a stutter dial tone on the Cisco Unified IP Phone to notify users of new voice messages. Users hear a stutter dial tone whenever the phone goes off hook on a line on which a voice message was left.

You can configure AMWI for all the phones in a cluster or for only certain directory numbers. The directory-number-level configuration takes precedence over the cluster-wide configuration.

Audible Message Waiting Indicator Prerequisites

You can configure AMWI only on Cisco Unified IP Phone that are running phone firmware Release 8.3(1) or later.

Audible Message Waiting Indicator Configuration Task Flow

Before you begin

• Review Audible Message Waiting Indicator Prerequisites, on page 187.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Generate a Phone Feature List, on page 1</td>
</tr>
<tr>
<td></td>
<td>Generate a report to identify devices that support the Audible Message Waiting Indicator feature.</td>
</tr>
</tbody>
</table>
### Configure Audible Message Waiting Indicator Service Parameters

This procedure describes how to configure AMWI default setting for all the phones in a cluster.

**Before you begin**

Generate a Phone Feature List, on page 1

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong> Configure Audible Message Waiting Indicator Service Parameters, on page 188</td>
<td>Configure AMWI default setting for all phones in a cluster.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure Audible Message Waiting Indicator for a Directory Number, on page 188</td>
<td>Configure AMWI for a directory number that is associated to a device.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure Audible Message Waiting Indicator for a SIP Profile, on page 189</td>
<td>Configure AMWI for SIP profiles. Perform this procedure to configure AMWI for SIP phones.</td>
</tr>
</tbody>
</table>

### Configure Audible Message Waiting Indicator for a Directory Number

Follow these steps to configure AMWI for a directory number that is associated with a device.

**Note**

The AMWI setting on an individual directory number overrides the clusterwide setting.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Phone</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the <strong>Association</strong> section, click <strong>Add a new DN</strong>.</td>
</tr>
<tr>
<td></td>
<td>The <strong>Directory Number Configuration</strong> window appears.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Select the <strong>Audible Message Waiting Indicator Policy</strong>. Choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• Off</td>
</tr>
</tbody>
</table>
• **On**—When you select this option, the users will receive a stutter dial tone when the handset is off hook.

• **Default**—When you select this option, the phone uses the default that was set at the system level.

---

**Step 4**

Configure the remaining 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 Audible Message Waiting Indicator for a SIP Profile**

Follow these steps to configure Audible Message Waiting Indicator (AMWI) for a SIP profile.

**Note**

The AMWI setting on an individual SIP profile overrides the clusterwide setting.

**Procedure**

**Step 1**

From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**. The **Find and List SIP Profiles** window appears.

**Step 2**

Enter the search criteria to use and click **Find**. The window displays a list of SIP profiles that match the search criteria.

**Step 3**

Click the SIP profile that you want to update. The **SIP Profile Configuration** window appears.

**Step 4**

Check the **Stutter Message Waiting** check box to activate stutter dial tone when the phone is off hook and a message is waiting.

**Step 5**

Click **Save**.

**Step 6**

Click **Apply Config**.

---

**Audible Message Waiting Indicator Troubleshooting**

**Audible Message Waiting Indicator Is Not Heard on the Phone**

**Problem**

Phone does not play stutter dial tone to notify the user of new voice messages.

If the user uses an SCCP phone, check the following:

• Ensure that the phone firmware release is 8.3(1) or later.

• Check the AMWI setting for the phone and the line on which the user went off hook.

• Verify that the Cisco CallManager service is running on the server.
Check the sniffer trace between the phone and Unified Communications Manager. Make sure that the phone receives the StartTone message with tone type equal to 42.

If the user uses a SIP phone, check the following:

- Ensure that the phone firmware release is 8.3(1) or later.
- Check the line (directory number) configuration. The phone must display the settings such as `line1_msgWaitingAMWI:1`, `line2_msgWaitingAMWI:0`.
- Ensure that the `Stutter Message Waiting` check box is checked in the `SIP Profile Configuration` window in Cisco Unified CM Administration.

## Localized AMWI Tone Is Not Played in a Specific Locale

**Problem** The phone that is configured in a non-English locale does not play the localized tone.

**Solution** Check the following:

- From Cisco Unified CM Administration, verify the User Locale in the `Device Profile Configuration` window (`Device > Device Settings > Device Profile`).
- Make sure that the user resets the phone after changing the locale.
- Check `user/local/cm/tftp /<locale name>` directory and verify that the AMWI tone is defined in the localized `g3-tones.xml` file.
Immediate Divert Overview

The Immediate Divert feature is a Unified Communications Manager supplementary service that allows you to immediately divert a call to a voicemail system. When Immediate Divert diverts a call, the line becomes available to make or receive new calls. Access the Immediate Divert feature by using the iDivert or Divert softkey on the IP phone.

Immediate Divert provides the following functions:

• Diverts a call to a voicemail system in the following manner:
  • Legacy iDivert diverts the call to the voice mailbox of the party that invokes the iDivert feature.
  • Enhanced iDivert diverts the call to either the voice mailbox of the party that invokes the iDivert feature or to the voice mailbox of the original called party.

• Diverts inbound calls that are in the Call Offering, Call on Hold, or Call Active states.

• Diverts outbound calls in the Call Active or Call on Hold states.

Note: Although the Immediate Divert feature is not available to CTI applications, a CTI redirect operation exists that performs the same function as Immediate Divert. Application developers can use the CTI redirect operation to accomplish Immediate Divert.
Immediate Divert Prerequisites

- You must configure the voicemail profiles and hunt pilots.
  
  For information on how to configure voicemail profiles and hunt pilots, see System Configuration Guide for Cisco Unified Communications Manager.

- The following devices support Immediate Divert:
  
  - Voice-messaging systems such as Cisco Unity Connection that use the Skinny Client Control Protocol (SCCP).
  
  - QSIG devices (QSIG-enabled H.323 devices, MGCP PRI QSIG T1 gateways, and MGCP PRI QSIG E1 gateways), depending on the setting of the Use Legacy Immediate Divert and Allow QSIG During iDivert clusterwide service parameters.

- The following table lists the phones that use the Divert or iDivert softkey.

<table>
<thead>
<tr>
<th>Cisco Unified IP Phone Model</th>
<th>Divert Softkey</th>
<th>iDivert Softkey</th>
<th>What to configure in softkey template</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 6900 Series (except 6901 and 6911)</td>
<td>X</td>
<td>X</td>
<td>iDivert</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7900 Series</td>
<td>X</td>
<td></td>
<td>iDivert</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 8900 Series</td>
<td>X</td>
<td></td>
<td>Configured by default</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 9900 Series</td>
<td>X</td>
<td></td>
<td>Configured by default</td>
</tr>
</tbody>
</table>

Note: Cisco Unified IP Phones 8900 and 9900 series have the Divert softkey assigned by default.

Immediate Divert Configuration Task Flow

Before you begin

- Review Immediate Divert Prerequisites, on page 192.
Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Immediate Divert Service Parameters, on page 193</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure a Softkey Template for Immediate Divert, on page 194</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>To Associate a Softkey Template with a Common Device Configuration, on page 195, complete the following subtasks: • Add a Softkey Template to the Common Device Configuration, on page 196 • Associate a Common Device Configuration with a Phone, on page 196</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Associate a Softkey Template with a Phone, on page 197</td>
</tr>
</tbody>
</table>

**Optional.** 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.** 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.

### Configure Immediate Divert Service Parameters

Procedure

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

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

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

**Step 4** Configure the relevant service parameters and click Save.

**Table 18: Service Parameter Fields for Immediate Divert**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park Display Timer</td>
<td>Enter a number from 0 to 100 (inclusive) to control the timer for the Immediate Divert text display on the IP phones. Set this timer for the server or for each server in a cluster that has the Cisco CallManager service and Immediate Divert configured. The default value for this service parameter is 10 seconds.</td>
</tr>
</tbody>
</table>
### Configure a Softkey Template for Immediate Divert

To divert incoming calls or outgoing calls, configure a softkey template and assign the iDivert softkey to that template. You can configure the iDivert softkey in the following call states:

- Connected
- On hold
- Ring in

Immediate Divert supports the following call states:

- For incoming calls:
  - Call offering (shown as Ring In on the softkey template).
  - Call on hold
  - Call active

- For outgoing calls:
  - Call on hold
  - Call active

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Legacy Immediate Divert</td>
<td>Select one of the following options from the drop-down list:</td>
</tr>
<tr>
<td></td>
<td>- <strong>True</strong>—The user that invokes the iDivert feature can divert an incoming call only to his own voice mailbox. This is the default setting.</td>
</tr>
<tr>
<td></td>
<td>- <strong>False</strong>—Immediate Divert allows diversion of an incoming call to either the voice mailbox of the original called party or to the voice mailbox of the user that invokes the iDivert feature.</td>
</tr>
<tr>
<td>Allow QSIG During iDivert</td>
<td>Select one of the following options from the drop-down list:</td>
</tr>
<tr>
<td></td>
<td>- <strong>True</strong>—Immediate Divert diverts calls to voicemail systems that can be reached over QSIG, SIP, and QSIG-enabled H.323 devices.</td>
</tr>
<tr>
<td></td>
<td>- <strong>False</strong>—Immediate Divert does not support access to voicemail systems over QSIG or SIP trunks. This is the default setting.</td>
</tr>
<tr>
<td>Immediate Divert User Response Timer</td>
<td>Enter a number from 5 to 30 (inclusive) to determine the time given to the iDivert softkey user to choose the party to whom to divert a call. If the user does not choose a party, the call remains connected. The default value for this service parameter is 5 seconds.</td>
</tr>
</tbody>
</table>
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:
   - If you modified a template that is already associated with devices, click **Apply Config** 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.

---

**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 *Associate a Softkey Template with a Phone, on page 197*

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add a Softkey Template to the Common Device Configuration, on page 196</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Common Device Configuration with a Phone, on page 196</td>
</tr>
</tbody>
</table>

**Add a Softkey Template to the 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.

**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**.
**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.

**Before you begin**

*Configure a Softkey Template for Immediate Divert, on page 194*

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Find** to select the phone to add the softkey template.
3. From the **Softkey Template** drop-down list, choose the template that contains the new softkey.
4. Click **Save**.

---

**Immediate Divert Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multilevel Precedence and Preemption (MLPP)</td>
<td>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) gets deactivated.</td>
</tr>
<tr>
<td>Call Forward</td>
<td>When the Forward No Answer setting on the Directory Number Configuration window is not configured, Call Forward uses the clusterwide CFNA timer service parameter, Forward No Answer Timer. If a user presses the iDivert softkey at the same time as the call is being forwarded, the call gets diverted to an assigned call forward directory number (because the timer was too short), not the voice-messaging mailbox. To resolve this situation, set the CFNA timer service parameter to enough time (for example, 60 seconds).</td>
</tr>
<tr>
<td>Call Detail Records (CDR)</td>
<td>Immediate Divert uses the immediate divert code number in the Onbehalf of fields (for example, joinOnbehalfOf and lastRedirectRedirectOnBehalfOf) in CDR.</td>
</tr>
</tbody>
</table>
When user A calls user B, and user B parks the call; user B retrieves the call and then decides to send the call to a voice-messaging mailbox by pressing the iDivert or Divert softkey. User A receives the voice-messaging mailbox greeting of user B.

When a conference participant presses the iDivert softkey, the remaining conference participants receive the voice-messaging mailbox greeting of the immediate divert initiator. Conference types include Ad Hoc, Meet-Me, Barge, eBarge, and Join.

For calls that reach the phone directly through a hunt list pilot (as part of the hunting algorithms), the iDivert softkey appears dimmed if the Use Legacy Immediate Divert clusterwide service parameter is set to True; otherwise, it does not appear dimmed.

For calls that do not reach the phone directly through a hunt list pilot (as part of the hunting algorithms), the iDivert softkey does not appear dimmed when the Use Legacy Immediate Divert clusterwide service parameter is set to True or False.

If the Use Legacy Immediate Divert clusterwide service parameter is set to False, and the Auto Call Pickup Enabled clusterwide service parameter is set to False, and a user of call pickup group uses call pickup to answer a call, the IP phone display will not present any choices to the user when the iDivert softkey is pressed.

### Immediate Divert Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Mail Profile</td>
<td>When you use QSIG integration with your voicemail system, a voicemail profile that includes either a voicemail pilot or a voicemail mask or both should leave the <strong>Make this the default Voice Mail Profile for the System</strong> check box unchecked. Ensure the default Voice Mail Profile setting is always set to No Voice Mail.</td>
</tr>
<tr>
<td>Call Forward All (CFA) and Call Forward Busy (CFB)</td>
<td>When Call Forward All (CFA) and Call Forward Busy (CFB) are activated, the system does not support Immediate Divert (CFA and CFB have precedence over Immediate Divert).</td>
</tr>
<tr>
<td><strong>Restriction</strong></td>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>---------------------------------</td>
<td>------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Busy Voicemail System</td>
<td>The iDivert detects a busy condition on the voicemail ports, when iDivert reaches a voicemail system over a local or SCCP connection.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Immediate Divert cannot divert a call to a busy voicemail port; voicemail ports can exist as members of a route or hunt list.</td>
</tr>
<tr>
<td></td>
<td>The call cannot divert to a busy voicemail system, but the original call gets maintained. The phone displays “Busy” message on which iDivert was invoked to indicate that the call was not diverted.</td>
</tr>
<tr>
<td></td>
<td>When a voicemail system is reached over a QSIG or SIP trunk, iDivert can be detected, but the call does not get maintained. When the <strong>Allow QSIG During iDivert clusterwide</strong> service parameter is set to <strong>True</strong>, or the <strong>Use Legacy Immediate Divert clusterwide</strong> service parameter is set to <strong>False</strong>, Immediate Divert supports access to voicemail systems that can be reached over QSIG or SIP trunks. When the <strong>Allow QSIG During iDivert clusterwide</strong> service parameter is set to <strong>False</strong>, and the <strong>Use Legacy Immediate Divert clusterwide</strong> service parameter is set to <strong>True</strong>, Immediate Divert does not support access to voicemail systems over QSIG or SIP trunks.</td>
</tr>
<tr>
<td>Malicious Caller ID</td>
<td>System does not support using Malicious Caller ID and Immediate Divert features together.</td>
</tr>
<tr>
<td>Forward No Answer Timeout</td>
<td>A race condition in connection with the Forward No Answer Timeout exists when you press the iDivert softkey. For example, if a manager presses the iDivert softkey immediately after the Forward No Answer timeout, call forward forwards the call to a preconfigured directory number. However, if the manager presses the iDivert softkey before the Forward No Answer timeout, immediate divert diverts the call to the voice-messaging mailbox of the manager.</td>
</tr>
<tr>
<td>Calling Parties and Called Parties</td>
<td>The calling parties and called parties can divert the call to their voice mailboxes if both simultaneously press the iDivert softkey.</td>
</tr>
<tr>
<td>Conference Types</td>
<td>When one participant in a conference presses the iDivert softkey, all remaining participants receive an outgoing greeting of the participant who pressed iDivert. Conference types include Meet-Me, Ad Hoc, cBarge, and Join.</td>
</tr>
<tr>
<td>Split or Join Operation</td>
<td>If the last action on a call was Auto Pickup, Call Transfer, Call Park, Call Park Reversion, Conference, Meet-Me Conference, or any application that performs a split or join operation, enhanced iDivert does not present a screen to a called party to choose the voice mailbox. Instead, enhanced iDivert immediately diverts the call to the voice mailbox that is associated with the called party.</td>
</tr>
</tbody>
</table>
Immediate Divert Troubleshooting

Key is not active

The phone displays this message when the user presses iDivert:

Key is not active

The voice-messaging profile of the user who pressed iDivert does not have a voice-messaging pilot. Configure a voice-messaging pilot in the user voice-messaging profile.

Temporary Failure

The phone displays this message when the user presses iDivert:

Temporary Failure

The voice-messaging system does not work, or a network problem exists. Troubleshoot your voice-messaging system. See troubleshooting or voice-messaging documentation.

Busy

The phone displays this message when the user presses iDivert:

Busy

This message means that the voice-messaging system is busy. Configure more voice-messaging ports or try again.
PART VII

Conferencing Features

• Ad Hoc Conferencing, on page 203
• Meet-Me Conferencing, on page 215
• Conference Now, on page 223
Ad Hoc Conferencing Overview

Ad Hoc conferences allow the conference controller (or in some cases, another participant) to add participants to the conference.

Ad Hoc conferences comprise two types: basic and advanced. In basic ad hoc conferencing, the originator of the conference acts as the controller of the conference and is the only participant who can add or remove other participants. In advanced Ad Hoc conferencing, any participant can add or remove other participants. Advanced Ad Hoc conferencing also allows you to link multiple ad hoc conferences together.

Advanced Ad Hoc conferencing allows you to link multiple Ad Hoc conferences together by adding an Ad Hoc conference to another Ad Hoc conference as if it were an individual participant. If you attempt to link multiple conferences together when the Advanced Ad Hoc Conference Enabled service parameter is set to False, the IP phone displays a message. You can also use the methods that are available for adding individual participants to an Ad Hoc conference to add another conference to an Ad Hoc conference.

Ad Hoc Conferencing 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 Softkey Template for Conferencing, on page 204</td>
<td>Add the Conference List, Join, and Remove Last Conference Party softkeys to a softkey template.</td>
</tr>
<tr>
<td>Step 2</td>
<td>To Associate Softkey Template Common Device, on page 205, complete the following subtasks:</td>
<td>Optional. To make the softkey template available to phones, you must complete either this step or the following step. Follow this step</td>
</tr>
</tbody>
</table>
Configure Softkey Template for Conferencing

Use this procedure to make the following conferencing softkeys available:

<table>
<thead>
<tr>
<th>Softkey</th>
<th>Description</th>
<th>Call States</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference List (ConfList)</td>
<td>View a list of participant directory numbers that are in an Ad Hoc conference. The name of the participant is displayed if it is configured in Cisco Unified Communications Manager Administration.</td>
<td>On Hook Connected</td>
</tr>
<tr>
<td>Join</td>
<td>Join up to 15 established calls (for a total of 16) to create a conference.</td>
<td>On Hold</td>
</tr>
<tr>
<td>Remove Last Conference Party (Remove)</td>
<td>The conference controller can invoke the conference list and remove any participant in the conference by using the Remove softkey.</td>
<td>On Hook Connected</td>
</tr>
</tbody>
</table>
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:
- If you modified a template that is already associated with devices, click Apply Config 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**
Complete one of the following procedures:
- Associate Softkey Template Common Device, on page 205
- Associate a Softkey Template with a Phone, on page 207

**Associate Softkey Template Common Device**

**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, go to **Associate a Softkey Template with a Phone, on page 207**

**Before you begin**

**Configure Softkey Template for Conferencing, on page 204**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add a Softkey Template to a Common Device Configuration, on page 206</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Common Device Configuration with a Phone, on page 207</td>
</tr>
</tbody>
</table>

**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**

1. From Cisco Unified CM Administration, choose Device > Phone.
2. Click Find and select the phone device to add the softkey template.
3. From the Common Device Configuration drop-down list, choose the common device configuration that contains the new softkey template.
4. Click Save.
5. Click Reset to update the phone settings.

**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**

1. From Cisco Unified CM Administration, choose Device > Phone.
2. Click Find to select the phone to add the softkey template.
3. From the Softkey Template drop-down list, choose the template that contains the new softkey.
4. Click Save.
5. Press Reset to update the phone settings.

**Configure Ad Hoc Conferencing**

Configure advanced Ad Hoc conferencing to allow non-controller participants to add and remove other participants and the ability of all participants to link ad hoc conferences together.

**Procedure**

1. From Cisco Unified CM Administration, choose System > Service Parameters.
2. From the Server drop-down list, choose the server.
3. From the Service drop-down list, choose Cisco CallManager.
4. Configure the fields in the Clusterwide Parameters (Features - Conference) area. For parameter descriptions, see Ad Hoc Conferencing Service Parameters, on page 208.
Step 5  
Click Save.

What to do next

Configure Join Across Lines, on page 210

Ad Hoc Conferencing Service Parameters

The following table lists the main service parameters for Ad Hoc conferencing. For additional conferencing service parameters, refer to the Service Parameter Configuration window's Advanced option. Conferencing service parameters appear under Clusterwide Parameters (Feature - Conference).

Table 19: Ad Hoc Conference Service Parameters

<table>
<thead>
<tr>
<th>Service Parameters</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Drop Ad Hoc Conference</td>
<td>Drop Ad Hoc Conference, prevents toll fraud (where an internal conference controller disconnects from the conference while outside callers remain connected). The service parameter settings specify conditions under which an ad hoc conference gets dropped.</td>
</tr>
<tr>
<td></td>
<td>• Never—The conference does not get dropped. (We recommend that you use the default option to avoid unintentional termination of a conference).</td>
</tr>
<tr>
<td></td>
<td>• When No OnNet Parties Remain in the Conference—The system drops the active conference when the last on-network party in the conference hangs up or drops out of the conference. Unified Communications Manager releases all resources that are assigned to the conference.</td>
</tr>
<tr>
<td></td>
<td>• When Conference Controller Leaves—The active conference terminates when the primary controller (conference creator) hangs up. Unified Communications Manager releases all resources that are assigned to the conference.</td>
</tr>
<tr>
<td></td>
<td>Note  We recommend that you set this service parameter to Never. Any other setting can result in unintentional termination of a conference.</td>
</tr>
<tr>
<td></td>
<td>The Drop Ad Hoc Conference service parameter works differently for conference calls that are initiated from a Cisco Unified IP Phone 7940 or 7960 that is running SIP, or a third-party phone that is running SIP.</td>
</tr>
<tr>
<td>Maximum Ad Hoc Conference</td>
<td>This parameter specifies the maximum number of participants that are allowed in a single Ad Hoc conference.</td>
</tr>
<tr>
<td></td>
<td>Default Value: 4</td>
</tr>
<tr>
<td>Service Parameters</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Advanced Ad Hoc Conference Enabled</td>
<td>This parameter determines whether advanced Ad Hoc conference features are enabled. This includes the ability of non-controller participants to add and remove other participants and the ability of all participants to link ad hoc conferences together.</td>
</tr>
<tr>
<td>Non-linear Ad Hoc Conference Linking Enabled</td>
<td>This parameter determines whether more than two Ad Hoc conferences can be linked directly to an Ad Hoc conference in a non-linear fashion (three or more conferences linked to any one conference).</td>
</tr>
</tbody>
</table>
Service Parameters | Description
--- | ---
Enable Click-to-Conference for Third-Party Applications | This parameter determines whether the Click-to-Conference functionality over the SIP trunk is enabled on Unified Communications Manager. The Click-to-Conference feature allows third-party applications to setup a conference using the SIP out of dialog REFER method and subscribe to the SIP trunk for Conference Event Package through SIP SUBSCRIBE/NOTIFY.
**Warning** Enabling this parameter could negatively affect CTI applications that are not coded to support this feature.
Default value: False

Cluster Conferencing Prefix Identifier | This parameter defines a number, up to 8 digits (e.g. 0001), that is prefixed to a conference identifier generated for Adhoc and Meet-Me conferences that will be hosted on a SIP conference bridge such as Cisco Telepresence MCU or Cisco Telepresence Conductor. This field should be populated by the administrator when there are multiple clusters in a network that will be sharing the SIP conference bridges that Unified Communications Manager manages. Every cluster should be configured with a unique prefix to ensure that the conference identifier for Adhoc and Meet-Me conferences is unique. If conference resources are not being shared across clusters, then this field may not be populated.

Configure Join Across Lines

The Join Across Lines feature allows a user to join calls on multiple phone lines (either on different directory numbers or on the same directory number but on different partitions) to create a conference.

**Before you begin**
- Ensure the phone model supports Join Across Lines [Generate a Phone Feature List, on page 1](#)
- Configure Ad Hoc Conferencing, on page 207

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > Device Settings > Default Device Profile. The Default Device Profile Configuration window is displayed.

**Step 2** From the Device Profile Type drop-down list, choose the phone model.

**Step 3** From the Device Protocol drop-down list, choose the relevant SCCP or SIP protocol.

**Step 4** Set the Join Across Lines to On.

**Step 5** Click Save.
Conference Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference by Using cBarge</td>
<td>Initiate a conference by pressing the cBarge softkey, or if the Single Button cBarge feature is enabled, by pressing the shared-line button of the active call. When cBarge is initiated, a barge call gets set up by using the shared conference bridge, if available. The original call gets split and then joined at the conference bridge. The call information for all parties gets changed to Conference. The barged call becomes a conference call with the barge target device as the conference controller. It can add more parties to the conference or can drop any party. When any party releases from the call, leaving only two parties in the conference, the remaining two parties experience a brief interruption and then get reconnected as a point-to-point call, which releases the shared conference resource.</td>
</tr>
<tr>
<td>Interaction with Call Park, Call Transfer, and Redirect</td>
<td>If the conference controller transfers, parks, or redirects the conference to another party, the party that retrieves the call acts as the virtual controller for the conference. A virtual controller cannot add new parties to the conference nor remove any party that was added to the conference, but a virtual controller can transfer, park, or redirect the conference to another party, who would, in turn, become the virtual controller of the conference. When this virtual controller hangs up the call, the conference ends.</td>
</tr>
<tr>
<td>Softkey display on SIP phones</td>
<td>The ConfList and the Remove softkey feature is available only on SCCP phones. The SIP phones have a Show Details button with similar functionality.</td>
</tr>
</tbody>
</table>

Conference Restrictions

The following restrictions apply to ad hoc conferencing:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ad Hoc conference</td>
<td>Unified Communications Manager supports a maximum of 100 simultaneous Ad Hoc conferences for each Unified Communications Manager server. Cisco Unified Communications Manager supports a maximum of 64 participants per Ad Hoc conference (provided adequate conference resources are available). In the case of linked Ad Hoc conferences, the system considers each conference as one participant.</td>
</tr>
</tbody>
</table>
## Conference Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ad Hoc conference on SIP phones:</td>
<td>Unified Communications Manager uses “beep” and “beep beep” tones when a new party is added and when the new party drops from the Ad Hoc conference, respectively. When a party is added to an Ad Hoc conference, a user on a phone that is running SIP may not hear the “beep beep”. Users might not hear the beeps because of the time it takes Unified Communications Manager to set up and tear down connections during the conferencing process. You can invoke Ad Hoc conference linking for phones that are running SIP only by using the Conference and Transfer functions. The system does not support Direct Transfer and Join. Supported phones that are running SIP comprise Cisco Unified IP Phone 7911, 7941, 7961.</td>
</tr>
<tr>
<td>• Cisco Unified IP Phone 7911</td>
<td></td>
</tr>
<tr>
<td>• Cisco Unified IP Phone 7941</td>
<td></td>
</tr>
<tr>
<td>• Cisco Unified IP Phone 7961</td>
<td></td>
</tr>
<tr>
<td>Ad Hoc conference on SIP phones:</td>
<td>• Phones display individual calls as conference calls. Cisco Unified IP Phones 7940 and 7960 can create local conference calls but not Ad Hoc conference calls.</td>
</tr>
<tr>
<td>• Cisco Unified IP Phone 7940</td>
<td>• Conference list (ConfList), is not available.</td>
</tr>
<tr>
<td>• Cisco Unified IP Phone 7960</td>
<td>• Remove last conference participant (RmLstC), is not available.</td>
</tr>
<tr>
<td>• Third-Party Phone</td>
<td>• Drop Ad Hoc conference is not supported.</td>
</tr>
<tr>
<td></td>
<td>• The SIP Profile parameter Conference Join Enabled controls behavior of the phone that is running SIP when the conference controller exits a locally hosted conference. If the Conference Join Enabled check box is unchecked, all legs disconnect when the conference controller exits the Ad Hoc conference call. If the Conference Join Enabled check box is checked, the remaining two parties stay connected.</td>
</tr>
<tr>
<td></td>
<td>• To achieve the same level of control that the Drop Ad Hoc Conference parameter settings provide for conference calls that a phone that is running SCCP initiates, the administrator can use a combination of the Conference Join Enabled SIP profile parameter and the Block OffNet to OffNet Transfer service parameter for conferences that are initiated on the phone that is running SIP (Cisco Unified IP Phone 7940 or 60). (Because the phone that is running SIP performs a transfer when it drops out of the conference call, the Block OffNet to OffNet Transfer can prevent toll fraud by not allowing two offnet phones to remain in the call.)</td>
</tr>
<tr>
<td></td>
<td>• Unified Communications Manager uses “beep” and “beep beep” tones when a new party is added and when the new party drops from the Ad Hoc conference, respectively. When a party is added to an Ad Hoc conference, a user on a phone that is running SIP may not hear the beep when a participant drops from the Ad Hoc conference, a user on a phone that is running SIP may not hear the “beep beep”. Users might not hear the beeps because of the time it takes Unified Communications Manager to set up and tear down connections during the conferencing process.</td>
</tr>
</tbody>
</table>
### Conference Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
</table>
| Phone displaying "To Conference" even when two parties are connected | Configure a Call Manager cluster with Publisher (CmA1) and Subscribers (CmA2). Phones A, B, C are registered with CmA1. Phones D is registered with CmA2.  
  - Setup an consultative or blind ad-hoc conference between A(1000), B(4000), C(5000), D(6000) with A as the controller.  
  - Shutdown CmA2.  
  - Phone D will go to Preservation mode & press end call softkey.  
  - Phone A,B & C are in conference.  
  - Phone A,B & C are in conference.  
  - Disconnect Phone A, then Phone B & C should be in a Direct call. Issue: Phone B & C are still in conference  
  - Disconnect Phone A, then Phone B & C should be in a Direct call. Issue: Phone B & C are still in conference  
  - Disconnect Phone B, there should be no call on phone C. Phone B & C are still in conference. Issue: Phone C is still in Conference. |
# Meet-Me Conferencing

Users can use Meet-Me Conferencing to set up or join conferences. A user that sets up a conference is called the conference controller. A user that joins a conference is called a participant.

## Meet-Me Conferencing Task Flow

### Before you begin

- Refer to the configuration documentation which came with your router and check for any settings which you may need to configure before proceeding with the Meet-Me Conferencing Task Flow.

### 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>Configure a Softkey Template for Meet-Me Conferencing, on page 216</td>
<td>Add the Meet-Me softkey to a softkey template.</td>
</tr>
</tbody>
</table>
| **Step 2** | To Associate a Softkey Template with a Common Device Configuration, on page 217, complete the following subtasks:  
  - Add a Softkey Template to a Common Device Configuration, on page 217  
  - Associate a Common Device Configuration with a Phone, on page 218 | **Optional.** To make the softkey template available to phones, you must complete either this step or the following step. |
| **Step 3** | Common Device Configuration Associate a Softkey Template with a Phone, on page 218 | **Optional.** Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in |
Configure a Softkey Template for Meet-Me Conferencing

Use this procedure to make the Meet Me softkey available in the off hook call state.

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:

- If you modified a template that is already associated with devices, click **Apply Config** 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.

---

**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 Associate a Softkey Template with a Phone, on page 218.

**Before you begin**

Configure a Softkey Template for Meet-Me Conferencing, on page 216

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add a Softkey Template to a Common Device Configuration, on page 217</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Common Device Configuration with a Phone, on page 218</td>
</tr>
</tbody>
</table>

---

**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.

---

**What to do next**

Associate a Common Device Configuration with a Phone, on page 218

### Associate a Common Device Configuration with a Phone

#### Before you begin

Add a Softkey Template to a Common Device Configuration, on page 217

#### 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.

---

**What to do next**

Configure a Meet-Me Conferencing Number, on page 219

### 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.

#### Before you begin

Configure a Softkey Template for Meet-Me Conferencing, on page 216
Configure a Meet-Me Conferencing Number

Configure a Meet-Me Conferencing Number

The Cisco Unified Communications Manager administrator provides the Meet-Me conference directory number range to users, so that they can access the feature. The user chooses a directory number from the range that is specified for the Meet-Me Number or Pattern to establish a Meet-Me conference and becomes the conference controller.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing > Meet-Me Number/Pattern**. The **Find and List Meet-Me Numbers** window appears.

Step 2 Enter the appropriate search criteria and click **Find**. All matching records are displayed.

Step 3 In the list of records, click the link for the record that you want to view.

Step 4 Perform one of the followings tasks:

- To copy a Meet-Me number or pattern, click the Meet-Me number or pattern that you want to copy. The **Meet-Me Number/Pattern Configuration** window appears. Click **Copy**.
- To add a Meet-Me Number or Pattern, click the **Add New** button.
- To update an existing Meet-Me Number or Pattern, click the Meet-Me Number or Pattern that you want to update.

Step 5 Enter the appropriate settings.

See the Related Topics section for information about the fields and their configuration options.

Step 6 Click **Save**.
# Meet-Me Number and Pattern Settings

<table>
<thead>
<tr>
<th><strong>Field</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory Number or Pattern</td>
<td>Enter a Meet-Me number or a range of numbers. To configure a range, the dash must appear within brackets and follow a digit; for example, to configure the range 1000 to 1050, enter 10[0-5]0.</td>
</tr>
<tr>
<td>Description</td>
<td>The description can include up to 50 characters in any language, but it cannot include double quotation marks (&quot;), percentage sign (%), ampersand (&amp;), or angle brackets (&lt;&gt;).</td>
</tr>
<tr>
<td>Partition</td>
<td>To use a partition to restrict access to the Meet-Me number or pattern, choose the desired partition from the drop-down list box. If you do not want to restrict access to the Meet-Me number or pattern, choose &lt;None&gt; for the partition. You can configure the number of partitions that are displayed in this drop-down list box by using the Max List Box Items enterprise parameter. If more partitions exist than the Max List Box Items enterprise parameter specifies, the Find button is displayed next to the drop-down list box. Click the Find button to display the Find and List Partitions window.</td>
</tr>
<tr>
<td>Minimum Security Level</td>
<td>Choose the minimum Meet-Me conference security level for this Meet-Me number or pattern from the drop-down list box. Choose Authenticated to block participants with nonsecure phones from joining the conference. Choose Encrypted to block participants with authenticated or nonsecure phones from joining the conference. Choose Non Secure to allow all participants to join the conference.</td>
</tr>
</tbody>
</table>

**Note** To set the maximum list box items, choose System > Enterprise Parameters and update the Max List Box Items field under CCMAdmin Parameters.

**Note** Make sure that the combination of Meet-Me number or pattern and partition is unique within the Unified Communications Manager cluster.

## Meet-Me Conferencing Restrictions

Unified Communications Manager supports a maximum of 100 simultaneous Meet-Me conferences for each Unified Communications Manager server.
After the maximum number of participants that is specified for that conference is has been exceeded, no other callers can join the conference.
Conference Now Overview

Conference Now provides a basic audio conferencing solution for small business customers that allows internal and external callers to join a conference via a centralized IVR.

To host a meeting, configured users must configure a meeting PIN that they will need to enter, along with the meeting number, when starting the meeting. The host provides the other meeting participants with the relevant meeting information, including the time slot, meeting number (which is usually the host’s primary extension) and an optional access code for secure conferences. At the designated time, the other participants can join the call by dialing the IVR, and entering the meeting information at the prompts.

Administrators must configure end users with the ability to host Conference Now conferences. After the feature is configured, meeting hosts can edit their meeting access code from within the Self-Care Portal.

Note

Cisco recommends that you use IPVMS software-based conference bridges for Conference Now. If you use other conference bridges, the conference entry and exit tones may not play to participants.

Conference Now Prerequisites

To use Conference Now you must make sure that the following media resources are configured, and are available to the devices that will be initiating conferences.
Conference Bridge—For the best user experience, we recommend using a software-based Cisco IPVMS conference bridge. Using another conference bridge might not provide the conference party entry and exit tone.

Interactive Voice Response (IVR)

After you configure these resources, you can make them available to devices by configuring a media resource group list that includes these resources and then associating that media resource group list to the device pools that will be used by your devices, or to individual devices. For more information on configuring Conference Bridges, Interactive Voice Response, and Media Resource Groups, see "Configure Media Resources" section of the System Configuration Guide for Cisco Unified Communications Manager.

Activate Cisco IP Voice Media Streaming

The Cisco IP Voice Media Streaming Service must be running in order to use IVR services and Conference Now.

Procedure

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

Step 2
From the Server drop-down list, choose the Cisco Unified Communications Manager publisher node.

Step 3
If the Cisco IP Voice Media Streaming application is deactivated, check the corresponding check box and click Save.

Configure Conference Now Settings

Use this procedure to configure Conference Now system settings on Unified Communications Manager.

Procedure

Step 1
From Cisco Unified CM Administration, choose Call Routing > Conference Now.

Step 2
In the Conference Now IVR Directory Number field, enter a DID (Direct Inward Dial) number for a Unified Communications Manager cluster to provide access for external callers.

Step 3
From the Route Partition drop-down list, select a partition.

Note The combination of the number and the partition must be unique within a cluster.

Step 4
Complete the remaining fields in the Conference Now Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 5
Click Save.
What to do next

Enable the feature for end users:

- If you haven't yet synced your LDAP directory, add Conference Now to your LDAP sync, so that newly synced users will be able to host Conference Now meetings. See Enable Conference Now via LDAP, on page 225.
- To enable the feature for an existing end user, see Enable Conference Now for User, on page 225.

Enable Conference Now for User

Use this procedure to configure an existing end user with the ability to host Conference Now meetings.

Note

You can also use Bulk Administration's Update Users feature to enable Conference Now for a large number of users via a csv file. You must ensure that the same settings as in the below task are configured. For more information, on how to use Update Users, see Bulk Administration Guide for Cisco Unified Communications Manager.

Procedure

Step 1
From Cisco Unified CM Administration, choose User Management > End User.

Step 2
Click Find and select the user for whom you want to add Conference Now.

Step 3
Under Conference Now, check the Enable End User to Host Conference Now check box.

Step 4
(Optional) For secure conferencing, enter an Attendees Access Code. Note that end users will be able to modify their access code setting within the Self-Care Portal.

Note
If the user has a Self-Service User ID assigned, the Conference Now Meeting Number prepopulates with the value of the Self-Service User ID, which defaults to the user's primary extension.

Step 5
Complete any remaining fields within the End User Configuration window. For more information on the fields and their configuration options, see the system Online Help.

Step 6
Click Save.

Enable Conference Now via LDAP

If you have not yet synced your LDAP directory, you can enable Conference Now for synced users by adding the option to a feature group template and then adding that feature group template to the initial LDAP sync. New users provisioned via the LDAP sync will have Conference Now enabled.

Note
You cannot apply feature group template edits to an LDAP directory sync where the initial sync has already occurred. To apply these edits to an LDAP sync, the initial sync must not yet have occurred.
Procedure

**Step 1**  
From Cisco Unified CM Administration, choose **User Management > User Phone/Add > Feature Group Template**.

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

**Step 3**  
From the drop-down list, select a **Service Profile**.

**Step 4**  
From the drop-down list, select a **UserProfile**.

**Step 5**  
Check the **Enable End User to Host Conference Now** check box.

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

What to do next

Assign the template to an LDAP directory sync so that synced users will be configured for Conference Now. For more information on configuring an LDAP sync, see the "Configure End Users" section of the System Configuration Guide for Cisco Unified Communications Manager.

Alternatively, you can add a new user with Conference Now functionality via the **Quick User/Phone Add** menu. You would need to add a new user that uses this feature group template in addition to assigning a primary extension.

**Conference Now Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions</th>
</tr>
</thead>
</table>
| Mobility EFA (Enterprise Feature Access) | A mobility user dials an Enterprise Feature Access DID number from a remote destination. After the call is connected, the remote destination phone is used to send DTMF digits to Unified Communications Manager via the PSTN gateway.  
The user PIN followed by the # key is first authenticated with Unified Communications Manager. After the user PIN authentication is successful, press 1 and the # key, to indicate a two-stage dialed call, followed by the desired phone number. If the dialed phone number is a Conference Now IVR Directory Number and the user is a meeting host, then the user must enter the PIN again.                                                                                     |
| Mobility MVA (Mobile Voice Access) | A call is directed to Unified Communications Manager through the enterprise PSTN H.323 or SIP gateway. The IVR prompts the user to enter the User ID, # key, PIN, # key, number 1 (to make a Mobile Voice Access call) and then the desired phone number. If the phone number is a Conference Now IVR Directory Number and the user is a meeting host, then the user must enter the PIN again. 
**Note**  
Users are not prompted for entering their PIN if they dial directly from their remote destination. However, if they dial from a different phone to Mobile Voice Access Directory Number, then they are prompted to enter PIN before they can make the call. If the users call Conference Now IVR Directory Number, they are prompted to enter the PIN again. |
Conference Now Restrictions

The Conference Now feature has the following restrictions:

- The host cannot mute attendees.
- The attendee cannot mute the audio by entering DTMF digits.
- The list of Conference Now participants is not supported.
- Maximum number of participants in a conference is controlled by the existing CallManager service parameter "Maximum MeetMe Conference Unicast". It applies to both internal and external callers.
- Maximum number of simultaneous Conference Now and MeetMe conference instances combined together is 100 per Unified Communications Manager CallManager node.
- Video on hold is not supported.
- The IPVMS software conference bridge only supports codec G.711 (ALaw & ULaw) and Wide Band 256k. If there is a codec mismatch between the calling device and the software conference bridge, a transcoder will be allocated.
- Ensure that at least one of the following conditions are met to play the conference party entry and exit tone:
  - At least one conference participant is using the Cisco IP Phone.
  - IPVMS is the allocated software conference bridge.
- When the sets up a Conference Bridge, the conference will continue with the remaining attendees irrespective whether the host is present or not. If the host wants to rejoin the conference, an announcement to enter the Attendee Access Code is played if it is configured by host. The host cannot schedule or mute attendees; therefore, the host status is no longer valid.
- No audio announcement will play if the host is the first person to join the conference. However, when the host dials into Conference Now from an internal IP Phone, there is a visual display on the IP Phone showing “To Conference”.

Note: If the host joins the Conference Now from any external phone, then there will be no visual display on the phone.
PART VIII

Placing Calls

- Call Back, on page 231
- Hotline, on page 243
- Speed Dial and Abbreviated Dial, on page 257
- WebDialer, on page 261
- Paging, on page 279
- Intercom, on page 301
Call Back

• Call Back Overview, on page 231
• Call Back Prerequisites, on page 231
• Call Back Configuration Task Flow, on page 232
• Call Back Interactions, on page 237
• Call Back Restrictions, on page 238
• Call Back Troubleshooting, on page 239

Call Back Overview

The CallBack feature allows you to receive notification when a busy extension is available to receive calls. You can activate Call Back for a destination phone that is within the same Unified Communications Manager cluster as your phone or on a remote Private Integrated Network Exchange (PINX) over QSIG trunks or QSIG-enabled intercluster trunks.

To receive CallBack notification, press the CallBack softkey or feature button while receiving a busy or ringback tone. You can activate Call Back during reorder tone, which is triggered when the No Answer timer expires.

Suspend/Resume

The Call Back feature enables the system to suspend the call completion service if the user who originated Call Back is busy. When the originating user then becomes available, the call completion service resumes for that user.

Note

Call Back supports Suspend/Resume CallBack notification for both intracluster and intercluster QSIG trunks or QSIG-enabled intercluster trunks.

Call Back Prerequisites

To use the Call Back feature, the destination phone must be in one of the following locations:

• In the same Unified Communications Manager cluster as the user phone
• On a remote PINX over QSIG trunks
• On a remote PINX over QSIG-enabled intercluster trunks

If you want to use non-English phone locales or country-specific tones, you must install locales.

• The following devices support the Call Back feature:
  • Cisco Unified IP Phones 6900, 7900, 8900, and 9900 Series (except 6901 and 6911)
  • Cisco IP Phones 7800 and 8800 Series
  • Cisco VGC Phone (uses the Cisco VG248 Gateway)
  • Cisco Analog Telephone Adapter (ATA) 186 and 188
  • Busy Subscriber for Cisco VG224 endpoints
  • No Answer for Cisco VG224 endpoints

• A CTI route point that forwards calls to any of the supported phones.

Call Back Configuration Task Flow

Complete one of the task flows depending on whether your phone supports softkey or buttons.

Use this table to determine whether to configure the CallBack softkey or the button for the Call Back supported IP phones.

Table 20: Cisco IP Phones That Use CallBack Softkeys and Buttons

<table>
<thead>
<tr>
<th>Cisco Phone Model</th>
<th>CallBack Softkey</th>
<th>CallBack Button</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phone 6900 Series (except 6901 and 6911)</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7900 Series</td>
<td>X</td>
<td></td>
</tr>
<tr>
<td>Cisco IP Phone 7800 and 8800 Series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 8900 Series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 9900 Series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco IP Communicator</td>
<td>X</td>
<td></td>
</tr>
</tbody>
</table>

Before you begin

• Review Call Back Prerequisites, on page 231.
**Configure Softkey Template for CallBack**

CallBack softkey has the following call states:

- On Hook
- Ring Out
- Connected Transfer

Use this procedure to make the CallBack softkey available:

**Before you begin**

Ensure your phone supports Call Back.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Configure Softkey Template for CallBack, on page 233</strong></td>
<td>Perform this step to add CallBack softkey to template and configure the soft key using the Common Device Configuration or phone.</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Configure CallBack Button, on page 236</strong></td>
<td>Perform this step to add and configure the CallBack button to a phone.</td>
<td></td>
</tr>
</tbody>
</table>

**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:

- If you modified a template that is already associated with devices, click Apply Config 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 the following procedures:

- Associate CallBack Softkey Template with a Common Device Configuration, on page 234
- Associate CallBack Softkey Template with Phone, on page 235

Associate CallBack 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 Associate CallBack Softkey Template with Phone, on page 235.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add CallBack Softkey Template to the Common Device Configuration, on page 234</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Common Device Configuration with a Phone, on page 235</td>
</tr>
</tbody>
</table>

Add CallBack Softkey Template to the 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.

What to do next

Associate a Common Device Configuration with a Phone, on page 235

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.
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.

Associate CallBack Softkey Template with Phone

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 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.
Configure CallBack Button

The procedures in this section describe how to configure the CallBack button.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Phone Button Template for Call Back, on page 236</td>
<td>Perform this step to assign CallBack button features to line or speed dial keys.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate a Button Template with a Phone, on page 237</td>
<td>Perform this step to configure the CallBack button for a phone.</td>
</tr>
</tbody>
</table>

Configure Phone Button Template for Call Back

Follow this procedure when you want to assign features to line or speed dial keys.

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:

   • If you modified a template that is already associated with devices, click Apply Config to restart the devices.
   • If you created a new softkey template, associate the template with the devices and then restart them.
What to do next
Associate a Button Template with a Phone, on page 237

Associate a Button Template with a Phone

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.

Call Back Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forward</td>
<td>Calls that are made from CallBack notification screen will override all the Call Forward configured values on the target DN. The calls should be made before CallBack recall timer expires otherwise the calls will not override the Call Forward configured values.</td>
</tr>
<tr>
<td>CallBack notification with phones running SIP</td>
<td>CallBack notification works differently only for Cisco Unified IP Phones 7960 and 7940. All other SIP phones and all SCCP phones support on-hook and off-hook notification. The only way that Unified Communications Manager knows when a line on a SIP 7960 or 7940 phone becomes available is by monitoring an incoming SIP INVITE message that Unified Communications Manager receives from the phone. After the phone sends the SIP INVITE to Unified Communications Manager and the phone goes on-hook, Unified Communications Manager sends an audio and CallBack notification screen to the Cisco Unified IP Phone 7960 and 7940 (SIP) user.</td>
</tr>
</tbody>
</table>
Interaction Feature

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb (DND)</td>
<td>CallBack would work normally in case or when <strong>DND-Reject</strong> is set to <strong>Off</strong> at the originating or the terminating end. The behavior differs only when <strong>DND-Reject</strong> is set to <strong>On</strong>.</td>
</tr>
<tr>
<td></td>
<td>- <strong>DND-Reject On on Originating end</strong>—User A calls User B and invokes Call Back. User A goes on DND-R. After User B is available, the CallBack notification will still be displayed to User A. That is, user will still be notified with the availability of the other party irrespective of the DND status.</td>
</tr>
<tr>
<td></td>
<td>- <strong>DND-Reject On on Terminating end</strong>—User A calls User B, and User B has set <strong>DND-Reject to On</strong>. User A will get a fast busy tone. User A can initiate CallBack on a busy endpoint. If User B is still on DND-Reject and goes Offhook and Onhook, User A will get a notification “User B is available now but on DND-R”, and it will not show the Dial option. If User A does not choose to cancel, CallBack will still monitor User B until User B sets <strong>DND-Reject to Off</strong>.</td>
</tr>
<tr>
<td>Cisco Extension Mobility</td>
<td>When a Cisco Extension Mobility user logs in or logs out, any active call completion that is associated with Call Back is automatically canceled. If a called phone is removed from the system after Call Back is activated on the phone, the caller receives a reorder tone after pressing the Dial softkey. The user may cancel or reactivate Call Back.</td>
</tr>
</tbody>
</table>

## Call Back Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Back with video across CUBE</td>
<td>The Call Back feature does not work for video calls when the call is placed between two Unified CM clusters that are connected via CUBE with qsig-enabled SIP trunks. For additional detail, see CSCun46243.</td>
</tr>
<tr>
<td>SIP Trunks</td>
<td>Call Back is not supported over SIP trunks but is supported over QSIG-enabled SIP trunks.</td>
</tr>
<tr>
<td>Supported characters for name or number of calling or called party</td>
<td>Call Back only supports spaces and digits 0 through 9 for the name or number of the calling or called party. To work with CallBack, the name or number of the calling or called party cannot contain a pound sign (#) or asterisk (*).</td>
</tr>
<tr>
<td>Voicemail</td>
<td>You cannot activate Call Back if you forward all calls to Voice-Messaging System.</td>
</tr>
</tbody>
</table>
Call Back Troubleshooting

This section describes the problems, possible causes, and solutions for various scenarios, and error messages that are displayed on the IP phone for Call Back.

Unplug/Reset Phone After Pressing CallBack Softkey but Before CallBack Occurs

**Problem**
You have unplugged or reset the phone after pressing the CallBack Softkey but before activating CallBack.

**Possible Cause**
Unified Communications Manager cancels the Call Back activation.

**Solution**
After the caller phone registers, the caller phone does not display the Call Back activation window after the reset. The caller must press the CallBack Softkey to view the active Call Back service. CallBack notification occurs on the phone.

Caller Misses to View Availability Notification Before Phone Reset

**Problem**
In an intracluster or intercluster Call Back scenario, a caller initiates Call Back for a user, for example, User B, who is unavailable. When User B becomes available, the availability notification screen displays on the caller phone, and a tone plays. The caller misses the availability notification for some reason, and the phone resets.

The caller contacts a different user, User C, for example, and presses the CallBack softkey because User C appears busy. The replace/retain screen displays on the caller phone, but the screen does not state that the availability notification already occurred for User B.

**Possible Cause**
The user reset the phone.

**Solution**
After a phone reset but not during an active call, review the Call Back notifications on the phone. Press the CallBack softkey.

Call Back Error Messages

The following section describes the error messages that display on the IP phone screen.
CallBack Is Not Active

**Problem**
The following error message is displayed:

```
CallBack is not active. Press Exit to quit this screen.
```

**Possible Cause**
User pressed the CallBack softkey during the idle state.

**Solution**
Follow the recommended action provided in the error message.

CallBack Is Already Active

**Problem**
The following error message is displayed:

```
CallBack is already active on xxxx. Press OK to activate on yyyy. Press Exit to quit this screen.
```

**Possible Cause**
A user tried to activate Call Back, but it is already active.

**Problem**
Follow the recommended action provided in the error message.

CallBack Cannot Be Activated

**Problem**
The following error message is displayed:

```
CallBack cannot be activated for xxxx.
```

**Possible Cause**
When a user tried to activate Call Back, either the extension is not available in Unified Communications Manager database or there is no QSIG route to the destination (that is, the extension belongs to remote Proxy which is connected via non-QSIG trunk), and the extension is not found in the database.

**Solution**
The user must try again, or the administrator must add the directory number to the Cisco Unified CM Administration.
Key Not Active

Problem
During a call, the CallBack softkey displays on the phone and the user presses the CallBack softkey before the phone rings. But, the following error message is displayed on the phone:

Key Not Active

Possible Cause
User may not be pressing the CallBack softkey at the appropriate time.

Solution
Users must press the CallBack softkey after a ringing or busy signal is received. Pressing the softkey at the wrong time may cause an error message to display on the phone.
Hotline Overview

The Hotline feature extends the Private Line Automatic Ringdown (PLAR) feature, which allows you to configure a phone so that when the user goes off hook (or the NewCall softkey or line key gets pressed), the phone immediately dials a preconfigured number. The phone user cannot dial any other number from a phone that is configured for PLAR. Hotline adds the following additional restrictions and administrator controls for phones that use PLAR:

- Hotline devices (devices configured to use hotline) that receive calls will receive calls only from other hotline devices, and will reject non-hotline callers.
- You can configure a Hotline phone to call only, receive only, or both call and receive.
- You can restrict the features available on a Hotline phone by applying a softkey template to the phone.
- Analog hotline phones ignore inbound hookflash signals.

Route Class Signaling

Hotline uses route class signaling to allow Hotline phones to receive calls only from other Hotline phones. A route class is a DSN code that identifies the class of traffic for a call. The route class informs downstream devices about special routing or termination requirements. A Hotline phone can only accept calls from a Hotline phone with the same route class.

Call Screening

Hotline also provides Configurable Call Screening based on caller ID. Configurable Call Screening allows a receiving Hotline phone to screen calls based on caller ID information and allow only callers in a screening list to connect.
System Requirements for Hotline

The following hotline system requirements exist for Unified Communications Manager:

- Unified Communications Manager 8.0(1) or higher on each server in the cluster
- MGCP gateway POTS phones (FXS).
- SCCP gateway POTS phones (FXS).

Tip
Cisco Feature Navigator allows you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://cfn.cloudapps.cisco.com/ITDIT/CFN/.
You do not need a Cisco.com account to access Cisco Feature Navigator.

Hotline 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>Generate a Phone Feature List, on page 1</td>
<td>Log in to Cisco Unified Reporting and run a phone feature list report to determine which phones support Hotline.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Create Custom Softkey Template, on page 244</td>
<td>Optional. If you want to restrict features on a Hotline phone, create a softkey template that allows only the features that you want.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Hotline on Phones, on page 245</td>
<td>Enable the phone as a Hotline device.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure Route Class Signaling Task Flow, on page 246</td>
<td>Configure route class signaling to support the Hotline feature.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure Hotline to Call Only or Receive Only Task Flow, on page 250</td>
<td>Optional. If you want to restrict a Hotline phone to either originating calls only or terminating calls only, configure call and receive settings.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure Call Screening with a Calling Search Space, on page 252</td>
<td>Optional. Use calling search spaces and partitions to configure a call screening list for your Hotline phones.</td>
</tr>
</tbody>
</table>

Create Custom Softkey Template

When configuring Hotline, you can customize a softkey template to display only those features that you want to make available to a Hotline phone.
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.

**Before you begin**
Generate a Phone Feature List, on page 1

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Choose <strong>Device &gt; Device Settings &gt; Softkey Template</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Add New</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the drop-down list, select a softkey template and click <strong>Copy</strong> to create a new template.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Softkey Template Name</strong> field, enter a unique name to identify the softkey template.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Enter a description that describes the use of the template. The description can include up to 50 characters in any language, but it cannot include double-quotes (&quot;), percentage sign (%), ampersand (&amp;), backslash (), or angle brackets (&lt;&gt;).</td>
</tr>
<tr>
<td>Step 6</td>
<td>To designate this softkey template as the standard softkey template, check the <strong>Default Softkey Template</strong> check box.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>If you designate a softkey template as the default softkey template, you will not be able to delete this softkey template unless you first remove the default designation.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td></td>
<td>The softkey template gets copied, and the <strong>Softkey Template Configuration</strong> window redisplayes.</td>
</tr>
<tr>
<td>Step 8</td>
<td>(Optional) Click the <strong>Add Application</strong> button.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Configure the positions of the softkeys on the Cisco Unified IP Phone LCD screen.</td>
</tr>
<tr>
<td>Step 10</td>
<td>To save your configuration, click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**Configure Hotline on Phones**

Use this procedure to enable the phone as a Hotline device.

**Before you begin**
Optional. If you want to create a custom softkey template to display only those features that you want to make available to a Hotline phone, see Create Custom Softkey Template, on page 244.

**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 <strong>Device &gt; Phone</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> and select the phone that you want to enable as a Hotline device.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Check the <strong>Hotline Device</strong> check box.</td>
</tr>
</tbody>
</table>
**Configure Route Class Signaling Task Flow**

Perform this task flow to configure route class signaling for Hotline calls.

**Procedure**

<table>
<thead>
<tr>
<th></th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Enable Route Class Signaling in the Cluster, on page 246</td>
<td>Set the route class signaling clusterwide defaults for trunks and gateways to enabled.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong></td>
<td>The settings for individual trunks and gateways override the clusterwide defaults. If you use this service parameter to enable route class signaling across the cluster, route class signaling can still be disabled on an individual trunk or gateway.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Enable Route Class Signaling on Trunks, on page 247</td>
<td>Enable route class signaling on an individual trunk.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Enable Route Class Signaling on Gateways, on page 247</td>
<td>Enable route class signaling on an MGCP T1/CAS or MGCP PRI gateway.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Signaling Labels for the Hotline Route Class, on page 248</td>
<td>Configure SIP signaling labels for Hotline route classes.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure the Route Class on Hotline Route Patterns, on page 249</td>
<td>Configure the route class on the route patterns that are routing your Hotline calls.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Configure the Route Class on Hotline Translation Patterns, on page 249</td>
<td><strong>Optional.</strong> If you use translation patterns on your Hotline calls, configure the route class on your translation patterns.</td>
</tr>
</tbody>
</table>

**Enable Route Class Signaling in the Cluster**

When you set the Route Class Trunk Signaling Enabled service parameter to True, the default route class signaling setting for all trunks or gateways in the cluster that support route class signaling is set to enabled.
The settings for individual trunks and gateways override the clusterwide defaults. If you use this service parameter to enable route class signaling across the cluster, route class signaling can still be disabled on an individual trunk or gateway.

**Procedure**

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

**Step 2** Set the **Route Class Trunk Signaling Enabled** service parameter to **True**.

**Step 3** Click **Save**.

**What to do next**

Use the following procedures to configure route class signaling on individual trunks or gateways.

- Enable Route Class Signaling on Trunks, on page 247
- Enable Route Class Signaling on Gateways, on page 247

**Enable Route Class Signaling on Trunks**

Use this procedure to enable route class signaling on an individual trunk. The configuration for individual trunks overrides the clusterwide service parameter setting.

**Before you begin**

Follow the Enable Route Class Signaling in the Cluster, on page 246 procedure to use a clusterwide service parameter to configure the default route class signaling settings for all trunks in the cluster.

**Procedure**

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

**Step 2** Click **Find** and select the SIP trunk on which you want to enable route class signaling.

**Step 3** From the **Route Class Signaling Enabled** drop-down list box, choose one of the following options:

- **Default**—This trunk uses the setting from the **Route Class Signaling Enabled** service parameter.
- **Off**—Route class signaling is disabled for this trunk.
- **On**—Route class signaling is enabled for this trunk.

**Step 4** Click **Save**.

**Enable Route Class Signaling on Gateways**

Use this procedure to enable route class signaling on an individual MGCP PRI or MGCP T1/CAS gateway. The configuration for individual gateways overrides the clusterwide service parameter setting.
**Before you begin**

Follow the Enable Route Class Signaling in the Cluster, on page 246 procedure to use a clusterwide service parameter to set the default route class signaling setting for gateways in the cluster.

Perform the Enable Route Class Signaling on Trunks, on page 247 procedure to configure route class signaling for individual trunks.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Gateways</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> and select the gateway on which you want to configure route class signaling.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Route Class Signaling Enabled</strong> drop-down list box, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Default</strong>—This gateway uses the setting from the clusterwide Route Class Signaling Enabled service parameter.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Off</strong>—Route class signaling is disabled on this gateway.</td>
</tr>
<tr>
<td></td>
<td>• <strong>On</strong>—Route class signaling is enabled on this gateway.</td>
</tr>
<tr>
<td>Step 4</td>
<td>If you want to encode voice route class for voice calls, check the <strong>Encode Voice Route Class</strong> check box.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**Configure Signaling Labels for the Hotline Route Class**

You must configure a SIP signaling label value for the Hotline route class that you want to use.

**Before you begin**

Enable route class signaling on your trunks and gateways. For details, see Enable Route Class Signaling in the Cluster, on page 246.

**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>From the <strong>Server</strong> drop-down list, choose the server on which the CallManager service is running.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Service</strong> drop-down list, choose <strong>Cisco CallManager</strong>.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Advanced</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the <strong>SIP Route Class Naming Authority</strong> service parameter field, enter a value to represent the naming authority and context for the labels used in SIP signaling to represent route class. The default value is <strong>cisco.com</strong>.</td>
</tr>
<tr>
<td>Step 6</td>
<td>In the <strong>SIP Hotline Voice Route Class Label</strong> service parameter field, enter a label to represent the Hotline Voice route class. The default value is <strong>hotline</strong>.</td>
</tr>
<tr>
<td>Step 7</td>
<td>In the <strong>SIP Hotline Data Route Class Label</strong> service parameter field, enter a label to represent the Hotline Data route class. The default value is <strong>ccdata</strong>.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>
Configure the Route Class on Hotline Route Patterns

This procedure describes call routing instructions that are specific to Hotline devices. For more information on how to configure route patterns and translation patterns in your network, see the System Configuration Guide for Cisco Unified Communications Manager.

For each route pattern that you expect to route a Hotline call, you must set the route class for that route pattern to **Hotline Voice** or **Hotline Data**.

**Before you begin**

Configure Signaling Labels for the Hotline Route Class, on page 248

Before you perform this procedure, it is expected that your network call routing is set up with route patterns.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Route/Hunt &gt; Route Patterns</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> to display a list of route patterns in your network.</td>
</tr>
</tbody>
</table>
| Step 3 | For each T1/CAS route pattern that is used to route a Hotline call:  
   a) From the **Find and List Route Patterns** window, select the route pattern.  
   b) From the **Route Class** drop-down list box, choose either **Hotline Voice** or **Hotline Data** as the route class for this route pattern.  
   c) Click **Save**. |

Configure the Route Class on Hotline Translation Patterns

**Before you begin**

Before you perform this procedure, it is expected that you have set up network call routing with route patterns and translation patterns.

Perform the Configure the Route Class on Hotline Route Patterns, on page 249 procedure.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Translation Pattern</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> to display the translation patterns in your cluster.</td>
</tr>
</tbody>
</table>
| Step 3 | For each translation pattern that you want to use on a Hotline number, perform the following steps:  
   a) From the **Route Class** drop-down list box, select either **Hotline Voice** or **Hotline Data**.  
   b) Click **Save**. |
## Configure Hotline to Call Only or Receive Only Task Flow

The configuration example in this task flow describes how to set up a Hotline phone to either place calls only or receive calls only.

**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>Configure Partitions for Hotline Call Only Receive Only, on page 250</td>
<td>Create two partitions: one should be empty and the other will be assigned to a new CSS.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Calling Search Space for Hotline Call Only Receive Only, on page 250</td>
<td>Create a new calling search space and assign one of the new partitions to this CSS. This CSS will contain no other partition.</td>
</tr>
</tbody>
</table>
| **Step 3** | Perform one of the following procedures:  
  - Configure Call Only on Hotline Phone, on page 251  
  - Configure Receive Only on Hotline Phone, on page 251 | If you want to configure call only, assign the empty partition to the phone line. If you want to configure receive only, assign the new CSS to the phone. |

## Configure Partitions for Hotline Call Only Receive Only

If you want to configure a Hotline phone to either place calls only, or to receive calls only you must create two partitions.

**Procedure**

1. From Cisco Unified CM Administration, choose **Call Routing > Class of Control > Partitions**.
2. Click **Add New**.
3. Create a new partition.
4. Enter a unique name and description for the partition. For example, **IsolatedPartition**.
   - **Note**: This partition will not be assigned to any CSS.
5. Click **Save**
6. Repeat steps 2-5 and create a second partition. For example, **EmptyPartition**.
   - **Note**: This partition will not be assigned to any phone line, but it will be assigned to the NoRouteCSS.

## Configure Calling Search Space for Hotline Call Only Receive Only

You must create a calling search and assign one of the two partitions that you've created to the calling search space.
Before you begin
Configure Partitions for Hotline Call Only Receive Only, on page 250

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Calling Search Space.
Step 2 Click Add New.
Step 3 Enter a Name and Description for the calling search space.
Step 4 From the Available Partitions list box, use the arrows to select the Empty Partition partition.
Note Make sure that the partition is assigned to only this calling search space and to no phone lines.
Step 5 Click Save

What to do next
Perform one of the following procedures:
• Configure Call Only on Hotline Phone, on page 251
• Configure Receive Only on Hotline Phone, on page 251

Configure Call Only on Hotline Phone
If you have set up your partitions and calling search spaces, perform these steps to configure the Hotline phone to place calls only.

Before you begin
Configure Calling Search Space for Hotline Call Only Receive Only, on page 250

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Phone.
Step 2 Click Find and select the Hotline phone.
Step 3 From the left navigation pane, click the phone line. The Directory Number Configuration window displays.
Step 4 From the Route Partition drop-down list, select the empty partition that you created.
Step 5 Click Save.

Configure Receive Only on Hotline Phone
If you have created your calling search space and partitions already, perform these steps to configure the Hotline phone to receive calls only.
Configure Call Screening with a Calling Search Space

Configure call screening for any intraswitched (line to line) Hotline calls by assigning a unique CSS where the Hotline phones that are in the partitions are only those Hotline phones that you want to be able to call each other.

You can also configure call screening by creating translation patterns where each pattern matches each number pattern that you want to either allow or screen.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure Partitions for Hotline Call Screening, on page 252</td>
<td>Create any new partitions for your Hotline phone lines.</td>
</tr>
<tr>
<td>Step 2 Create Calling Search Space for Hotline Call Screening, on page 253</td>
<td>Create a new CSS for the screening list. The CSS must include partitions with only those Hotline numbers that you want to allow.</td>
</tr>
<tr>
<td>Step 3 Configure Hotline Phones for Call Screening, on page 254</td>
<td>Assign the new CSS and partition to the Hotline phone.</td>
</tr>
</tbody>
</table>

Configure Partitions for Hotline Call Screening

To configure call screening in Hotline phones using a calling search space, you must set up partitions where the only Hotline numbers are those that you want to allow.

Perform the following procedure if you need to create a new partition for your Hotline call screening list.

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. | |
Create Calling Search Space for Hotline Call Screening

Perform the following procedure to create a new calling search space for the Hotline phones in the call screening list. Make sure that the only Hotline numbers in the partitions that you select for this CSS are those Hotline numbers that you want to allow in the call screening list. No Hotline numbers that you want to screen out should be included in the partitions for this CSS.

Before you begin
Configure Partitions for Hotline Call Screening, on page 252

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 (<>).
Configure Hotline Phones for Call Screening

If you have already configured calling search spaces and partitions for Hotline call screening, perform this procedure to assign the calling search spaces and partitions to your Hotline phones.

**Before you begin**

Create Calling Search Space for Hotline Call Screening, on page 253

**Procedure**

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

**Step 2** Click Find and select the Hotline phone.

**Step 3** From the Calling Search Space drop-down list, select the new calling search space that you created for the Hotline call screening list.

**Step 4** Click Save.

**Step 5** From the left navigation pane, click the phoneline that you want to use for Hotline calls. The Directory Number Configuration window displays.

**Step 6** From the Route Partition drop-down list, select a partition that is included in the calling search space that you set up.

**Step 7** Click Save.

**Hotline Troubleshooting**

The following table provides troubleshooting information for cases where hotline calls do not dial correctly.

<table>
<thead>
<tr>
<th>Problem</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial tone</td>
<td>Check PLAR configuration.</td>
</tr>
</tbody>
</table>
| Reorder tone or VCA (intracluster call) | • Check PLAR configuration.  
                                        • Verify that the phones on both ends are configured as hotline phones. |
The following table provides troubleshooting information for cases where call screening based on caller ID does not work.

Table 22: Troubleshooting Hotline—Call Screening Based on Caller ID Problems

<table>
<thead>
<tr>
<th>Problem</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call not allowed</td>
<td>• Check Caller ID.</td>
</tr>
<tr>
<td></td>
<td>• Add pattern to screen CSS.</td>
</tr>
<tr>
<td>Call allowed</td>
<td>Remove pattern from screen CSS.</td>
</tr>
</tbody>
</table>

- Reorder tone or VCA (intercluster or TDM call)
  - Check PLAR configuration.
  - Verify that the phones on both ends are configured as hotline phones.
  - Verify that route class signalling is enabled on trunks.
  - Check the configuration of route class translations on CAS gateways.
Speed Dial and Abbreviated Dial

• Speed Dial and Abbreviated Dial Overview, on page 257
• Speed Dial and Abbreviated Dial Configuration Task Flow, on page 258

Speed Dial and Abbreviated Dial Overview

Administrators can configure speed dial numbers for phones to provide speed dial buttons for users or to configure phones that do not have a specific user that is assigned to them. Users use the Cisco Unified Communications Self Care Portal to change the speed dial buttons on their phones. When configuring speed dial entries, some of the speed dial entries are assigned to the speed dial buttons on the IP phone; the remaining speed dial entries are used for abbreviated dialing. When a user starts dialing digits, the AbbrDial softkey displays, and the user can access any speed dial entry by entering the appropriate index (code) for abbreviated dialing.

The speed dial settings on the phone are associated with a physical button on a phone, whereas the abbreviated dial settings are not associated with a phone button.

Programming Speed Dials with Pauses

You can program commas in your speed dials to reach destinations that require a Forced Authorization Code (FAC), Client Matter Code (CMC), dialing pause, or additional digits (such as a user extension, meeting access number, or voice mail password). Within a speed dial, each comma (,) represents either:

• A delimiter that separates the destination call address from an FAC or CMC code
• A pause of 2 seconds prior to sending post-connect DTMF digits

For example, let’s say that you want a speed dial that includes FAC and CMC codes, followed by IVR prompts where:

• The called number is 91886543.
• The FAC code is 8787.
• The CMC code is 5656.
• The IVR response is 987989#, which must be entered 4 seconds after the call connects.

In this case, you would program 91886543,8787,5656,,987989# as the speed dial.
## Speed Dial and Abbreviated Dial 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>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that support the Speed Dial and Abbreviated Dial feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Speed Dial and Abbreviated Dial, on page 258</td>
<td>Configure Speed Dial and Abbreviated Dial numbers.</td>
</tr>
</tbody>
</table>

### Configure Speed Dial and Abbreviated Dial

You can configure a total of 199 speed dial and abbreviated dial settings. Configure speed dial settings for the physical buttons on the phone. Configure abbreviated dial settings for the speed dial numbers that you access with abbreviated dialing. You can configure speed dial entries and abbreviated dial indexes in the same window.

You can also configure post connect DTMF digits as well as FAC, CMC codes as part of the speed dial.

Follow these steps to configure speed dial and abbreviated dial.

#### Note

Not all Cisco Unified IP Phones support abbreviated dialing. See the phone user guide for information.

#### Before you begin

Generate a Phone Feature List, on page 1

#### Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone**. Enter your search criteria and click **Find**. Choose the phone for which you want to configure speed dial buttons.

**Step 2** From the **Phone Configuration** window, choose **Add/Update Speed Dials** from the Related Links drop-down list at the top of the window and click **Go**.

The **Speed Dial and Abbreviated Dial Configuration** window appears for the phone.

**Step 3** In the **Number** field, enter the number that you want the system to dial when the user presses the speed dial button or the abbreviated dial index for abbreviated dial. You can enter digits 0 through 9, *, #, and +, which is the international escape character. To include dialing pauses in the speed dial, you can enter comma (,) which can act as a delimiter before sending DTMF digits. Each comma you include represents an additional pause of 2 seconds. For example, two commas (,,) represent a pause of 4 seconds. Use of commas also allows you to separate FAC and CMC from the other digits in the speed dial string.
**Note**  Ensure that the following requirements are met when you include FAC and CMC in the speed dial string:

- FAC must always precede CMC in the speed dial string.
- A speed dial label is required for speed dials with FAC and DTMF digits.
- Only one comma is allowed between FAC and CMC digits in the string.

**Step 4**  In the **Label** field, Enter the text that you want to display for the speed dial button or abbreviated dial number.

**Note**  This field is not available for all the phones. To determine whether this field is available for your Cisco Unified IP Phone, see the user documentation for your phone model.

**Step 5**  (Optional) If you are configuring a pause in speed dial, you must add a label so that FAC, CMC, and DTMF digits are not displayed on the phone screen.
Configure Speed Dial and Abbreviated Dial
WebDialer Overview

Cisco WebDialer is installed on a Unified Communications Manager node and used along with Unified Communications Manager. It allows Cisco Unified IP Phone users to make calls from web and desktop applications.

Cisco WebDialer uses hyperlinked telephone numbers in a company directory to allow users to make calls from a web page by clicking on the telephone number of the person that they are trying to call. Cisco WebDialer supports both IPv4 and IPv6 addressing.

In the Cisco Unified Communications Self-Care Portal, from the Directory window, launch Cisco WebDialer using a URL similar to the following:

https://<IP address of Cisco Unified Communications Manager server>:8443/webdialer/

WebDialer Prerequisites

Cisco WebDialer requires the following software components:

• CTI-supported Cisco Unified IP Phones

WebDialer Configuration Task Flow

Before you begin

• Review WebDialer Prerequisites, on page 261.
## 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 WebDialer, on page 263</td>
<td>Activate the WebDialer service.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>(Optional) Enable WebDialer Tracing, on page 263</td>
<td>To view WebDialer traces, enable tracing.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>(Optional) Configure WebDialer Servlet, on page 264</td>
<td>Configure the WebDialer servlet.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>(Optional) Configure Redirector Servlet, on page 264</td>
<td>If you have multi cluster applications that you develop using HTML over HTTPS interfaces, configure the Redirector servlet.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>(Optional) Configure WebDialer Application Server, on page 265</td>
<td>To configure Redirector for Cisco WebDialer.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>(Optional) To Configure Secure TLS Connection to CTI, on page 265, complete the following sub tasks:</td>
<td>WebDialer uses WDSecureSysUser application user credentials to establish a secure TLS connection to CTI to make calls. Follow these procedures if your system is running in mixed mode.</td>
</tr>
<tr>
<td></td>
<td>• Configure WDSecureSysUser Application User, on page 266</td>
<td>• Configure CAPF Profile, on page 161</td>
</tr>
<tr>
<td></td>
<td>• Configure Cisco WebDialer Web Service, on page 163</td>
<td>• Configure Cisco WebDialer Web Service, on page 163</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Configure Language Locale for WebDialer, on page 269</td>
<td>Determine which language WebDialer displays by setting the locale field in the Cisco Unified Communications Self Care Portal menu.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Configure WebDialer Alarms, on page 269</td>
<td>If there are any issues with the Web Dialer feature it alerts the administrator.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>(Optional) Configure Application Dial Rules, on page 270</td>
<td>If your application requires multiple clusters, configure application dial rules.</td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td>Add Users to Standard CCM End User Group, on page 271</td>
<td>Add each WebDialer user to the Standard End User Group for Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td><strong>Step 11</strong></td>
<td>(Optional) To Configure Proxy User, on page 271, complete the following sub tasks:</td>
<td>If you use makeCallProxy HTML over HTTP interface to develop an application for using Cisco WebDialer, create a proxy user.</td>
</tr>
<tr>
<td></td>
<td>• Add a WebDialer End User, on page 272</td>
<td>• Assign Authentication Proxy Rights, on page 272</td>
</tr>
</tbody>
</table>
Activate WebDialer

Procedure

Step 1  From Cisco Unified Serviceability, choose **Tools > Service Activation**.
Step 2  From the **Servers** drop-down list, choose the Unified Communications Manager server that is listed.
Step 3  From **CTI Services**, check the **Cisco WebDialer Web Service** check box.
Step 4  Click **Save**.
Step 5  From Cisco Unified Serviceability, choose **Tools > Control Center - Feature Services** to confirm that the CTI Manager service is active and is in start mode.

For WebDialer to function properly, the CTI Manager service must be active and in start mode.

What to do next

Configure Language Locale for WebDialer, on page 269 or complete any or all of the following optional tasks:

- Enable WebDialer Tracing, on page 263
- Configure WebDialer Servlet, on page 264
- Configure Redirector Servlet, on page 264
- Configure WebDialer Application Server, on page 265
- Configure Secure TLS Connection to CTI, on page 265

Enable WebDialer Tracing

To enable Cisco WebDialer tracing, use the Cisco Unified Serviceability Administration application. Trace settings apply to both the WebDialer and Redirector servlets. To collect traces, use the Real Time Monitoring Tool (RTMT).

To access the WebDialer trace files, use the following CLI commands:

- `file get activelog tomcat/logs/webdialer/log4j`
- `file get activelog tomcat/logs/redirector/log4j`

For more information about traces, see the *Cisco Unified Serviceability Administration Guide*.

Before you begin

Activate WebDialer, on page 263
**Configure WebDialer Servlet**

The WebDialer servlet is a Java servlet that allows Cisco Unified Communications Manager users in a specific cluster to make and complete calls.

**Before you begin**

*Activate WebDialer, on page 263*

**Procedure**

**Step 1** Choose System > Service Parameters.

**Step 2** From the Server drop-down list, choose the Cisco Unified Communications Manager server on which to configure Cisco WebDialer web service parameters.

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

**Step 4** Configure the relevant WebDialer Web Service parameters. For detailed information about the parameters, see online help.

**Step 5** Restart the Cisco WebDialer Web Service for new parameter values to take effect.

**Configure Redirector Servlet**

Redirector servlet is a Java-based Tomcat servlet. When a Cisco WebDialer user makes a request, the Redirector servlet looks for that request in the Cisco Unified Communications Manager cluster and redirects the request to the specific Cisco WebDialer server that is located in the Cisco Unified Communications Manager cluster. The Redirector servlet is available only for multi cluster applications that are developed by using HTML over HTTPS interfaces.
Configure WebDialer Application Server

Application server is required to configure the Redirector Servlet. Redirector is required only when you have multiple Unified Communications Manager servers configured in a cluster.

Before you begin

Activate WebDialer, on page 263

Procedure

Step 1 From Cisco Unified Communications Manager Administration Application server window, choose System > Application Server.
Step 2 From the Application Server Type drop-down list, choose a Cisco WebDialer application server. The server appears in the List of WebDialers field in the Service Parameter Configuration window for the Cisco WebDialer Web Service.

Configure Secure TLS Connection to CTI

WebDialer uses WDSecureSysUser application user credentials to establish a secure TLS connection to CTI to make calls. To configure the WDSecureSysUser application user to establish a secure TLS connection, complete the following tasks.

Before you begin

• Install and configure the Cisco CTL Client. For more information about CTL Client, see Security Guide for Cisco Unified Communications Manager.
• Verify that the Cluster Security Mode in the Enterprise Parameters Configuration window is 1 (mixed mode). Operating the system in mixed mode impacts other security functions in your system. If your system is not currently running in mixed mode, do not switch to mixed mode until you understand these interactions. For more information, see Security Guide for Cisco Unified Communications Manager.

• Verify that the Cluster SIPOAuth Mode field is set to Enabled.

• Activate the Cisco Certificate Authority Proxy Function service on the first node.

• Activate WebDialer, on page 263

---

**Configure WDSecureSysUser Application User**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure WDSecureSysUser Application User, on page 266</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure CAPF Profile, on page 161</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Cisco WebDialer Web Service, on page 163</td>
</tr>
</tbody>
</table>

---

**Configure WDSecureSysUser Application User**

**Procedure**

1. From Cisco Unified CM Administration, choose **User Management > Application User**.
2. Click **Find**.
3. From the **Find and List Application Users Application** window, choose **WDSecureSysUser**.
4. Configure the fields in the **Application User Configuration** window and click **Save**.

---

**What to do next**

Configure CAPF Profile, on page 161

---

**Configure CAPF Profile**

Certificate Authority Proxy Function (CAPF) is a component that performs tasks to issue and authenticate security certificates. When you create an application user CAPF profile, the profile uses the configuration details to open secure connections for the application.

**Procedure**

1. From Cisco Unified CM Administration, choose **User Management > Application User CAPF Profile**.
2. Perform one of the following tasks:
• To add a new CAPF profile, click Add New in the Find window.
• To copy an existing profile, locate the appropriate profile and click the Copy icon for that record in the Copy column.

To update an existing entry, locate and display the appropriate profile.

**Step 3**
Configure or update the relevant CAPF profile fields. See the Related Topics section information about the fields and their configuration options.

**Step 4**
Click Save.

**Step 5**
Repeat the procedure for each application and end user that you want to use security.

---

**CAPF Profile Settings**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application User</td>
<td>From the drop-down list, choose the application user for the CAPF operation. This setting displays configured application users.</td>
</tr>
<tr>
<td></td>
<td>This setting does not appear in the End User CAPF Profile window.</td>
</tr>
<tr>
<td>End User ID</td>
<td>From the drop-down list, choose the end user for the CAPF operation. This setting displays configured end users.</td>
</tr>
<tr>
<td></td>
<td>This setting does not appear in the Application User CAPF Profile window.</td>
</tr>
<tr>
<td>Instance ID</td>
<td>Enter 1 to 128 alphanumeric characters (a-z, A-Z, 0-9). The Instance ID identifies the user for the certificate operation.</td>
</tr>
<tr>
<td></td>
<td>You can configure multiple connections (instances) of an application. To secure the connection between the application and CTIManager, ensure that each instance that runs on the application PC (for end users) or server (for application users) has a unique certificate.</td>
</tr>
<tr>
<td></td>
<td>This field relates to the CAPF Profile Instance ID for Secure Connection to CTIManager service parameter that supports web services and applications.</td>
</tr>
<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—This message is displayed when no certificate operation is occurring. (default setting)</td>
</tr>
<tr>
<td></td>
<td>• Install/Upgrade—This option installs a new certificate or upgrades an existing locally significant certificate for the application.</td>
</tr>
<tr>
<td>Setting</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Authentication Mode</td>
<td>The authentication mode for the Install/Upgrade certificate operation specifies By Authentication String, which means CAPF installs, upgrades, or troubleshoots a locally significant certificate only when the user or administrator enters the CAPF authentication string in the JTAPI/TSP Preferences window.</td>
</tr>
<tr>
<td>Authentication String</td>
<td>To create your own authentication string, enter a unique string. Each string must contain 4 to 10 digits. To install or upgrade a locally significant certificate, the administrator must enter the authentication string in the JTAPI/TSP preferences GUI on the application PC. This string supports one-time use only; after you use the string for the instance, you cannot use it again.</td>
</tr>
<tr>
<td>Generate String</td>
<td>To automatically generate an authentication string, click this button. The 4- to 10-digit authentication string appears in the Authentication String field.</td>
</tr>
<tr>
<td>Key Size (bits)</td>
<td>From the drop-down list, choose the key size for the certificate. The default setting is 1024. The other option for key size is 512. Key generation, which is set at low priority, allows the application to function while the action occurs. Key generation may take up to 30 or more minutes.</td>
</tr>
<tr>
<td>Operation Completes by</td>
<td>This field, which supports all certificate operations, specifies the date and time by which you must complete the operation. The values that are displayed apply for the first node. Use this setting with the CAPF Operation Expires in (days) enterprise parameter, which specifies the default number of days in which the certificate operation must be completed. You can update this parameter at any time.</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
<td>This field displays the progress of the certificate operation, such as pending, failed, or successful. You cannot change the information that is displayed in this field.</td>
</tr>
</tbody>
</table>
Configure Cisco IP Manager Assistant

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 WebDialer Web service is active.
Step 3  From the Service drop-down list, choose the Cisco WebDialer Web service. A list of parameters appears.
Step 4  Navigate to and update the CTIManager Connection Security Flag and CAPF Profile Instance ID for Secure Connection to CTIManager parameters.
        To view parameter descriptions, click the parameter name link.
        Note  CTIManager supports IPv4 and IPv6 addresses.
Step 5  Click Save.
Step 6  Repeat the procedure on each server on which the service is active.

What to do next
Refer to the Manager Assistant Task Flow for Shared Lines, on page 153 to determine the next task to complete.

Configure Language Locale for WebDialer

Use the Cisco Unified Communications Self Care Portal to configure a language locale for Cisco WebDialer. The default language is English.

Before you begin
Activate WebDialer, on page 263

Procedure

Step 1  From the Cisco Unified Communications Self Care Portal, click the General Settings tab.
Step 2  Click Language.
Step 3  From the Display Language drop-down list, select a language local, and then click Save.

Configure WebDialer Alarms

Cisco WebDialer service uses Cisco Tomcat to generate alarms.

Before you begin
Configure Language Locale for WebDialer, on page 269
Procedure

**Step 1** In Cisco Unified Serviceability, choose **Alarm > Configuration**.

**Step 2** From the **Server** drop-down list, choose the server on which to configure the alarm and then click **Go**.

**Step 3** From the **Services Group** drop-down list, choose **Platform Services** and then click **Go**.

**Step 4** From the **Services** drop-down list, choose **Cisco Tomcat** and then click **Go**.

**Step 5** If your configuration supports clusters, check the **Apply to All Nodes** check box to apply the alarm configuration to all nodes in the cluster.

**Step 6** Configure the settings, as described in Alarm configuration settings, which includes descriptions for monitors and event levels.

**Note** For more information about the Alarm configuration settings, see the *Cisco Unified Serviceability Guide*.

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

**What to do next**

Add Users to Standard CCM End User Group, on page 271 or (optionally) if your application requires multiple clusters, see Configure Application Dial Rules, on page 270.

### Configure Application Dial Rules

**Before you begin**

Configure WebDialer Alarms, on page 269

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Call Routing > Dial Rules > Application Dial Rules**.

**Step 2** In the **Name** field, enter a name for the dial rule.

**Step 3** In the **Description** field, enter a description for the dial rule.

**Step 4** In the **Number Begins With** field, enter the initial digits of the directory numbers to which you want to apply this application dial rule.

**Step 5** In the **Number of Digits** field, enter the length of the dialed numbers to which you want to apply this application dial rule.

**Step 6** In the **Total Digits to be Removed** field, enter the number of digits that you want Cisco Unified Communications Manager to remove from the beginning of dialed numbers that apply to this dial rule.

**Step 7** In the **Prefix With Pattern** field, enter the pattern to prepend to dialed numbers that apply to this application dial rule.

**Step 8** For **Application Dial Rule Priority**, choose the dial rule priority as top, bottom, or middle.

**Step 9** Click **Save**.
Add Users to Standard CCM End User Group

To use the Cisco WebDialer links in the User Directory windows in Cisco Unified Communications Manager, you must add each user to the Standard Cisco Unified Communications Manager End Users Group.

Procedure

| Step 1 | Choose User Management > User Group. |
| Step 2 | In the Find and List User Group window, click Find. |
| Step 3 | Click Standard CCM End Users. |
| Step 4 | In the User Group Configuration window, click Add End Users to Group. |
| Step 5 | In the Find and List Users window, click Find. You can enter criteria for a specific user. |
| Step 6 | To add one or more users to the user group, complete one of the following steps: |
| | • To add one or more users, check the check box beside each user to add and then click Add Selected. |
| | • To add all users, click Select All and then click Add Selected. |

The users appear in the Users in Group table of the User Group Configuration window.

Configure Proxy User

If you use makeCallProxy HTML over HTTP interface to develop an application for using Cisco WebDialer, create a proxy user. For information about the makeCallProxy interface, see the makeCallProxy section in the Cisco WebDialer API Reference Guide.

Note

MakeCallProxy HTTP Methods is a service parameter under WebDialer Service. This parameter controls the HTTP methods that the MakeCallProxy API accepts. HTTP GET is considered insecure because the credentials required by the API are included as parameters in HTTP GET requests. Hence these HTTP GET parameters can be captured in the application logs and in the web browser's history.

When the service parameter MakeCallProxy HTTP Methods is set to Secure, request made by the HTTP GET will be rejected. By default the parameter MakeCallProxy HTTP Methods is set to Insecure, so that the API accepts both GET and POST methods and the backward compatibility is maintained.

Before you begin

Add Users to Standard CCM End User Group, on page 271
Add a WebDialer End User

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>(Optional) Add a WebDialer End User, on page 272</td>
<td>Add a new user. If the user exists, you can proceed to the next task.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign Authentication Proxy Rights, on page 272</td>
<td>Assign authentication proxy rights to an end user.</td>
</tr>
</tbody>
</table>

Add a WebDialer End User

Procedure

- **Step 1**: From Cisco Unified CM Administration, choose User Management > End User.
- **Step 2**: Click Add New.
- **Step 3**: Enter a Last Name.
- **Step 4**: Enter and confirm a Password.
- **Step 5**: Enter and confirm a PIN.
- **Step 6**: Complete any remaining fields in the End User Configuration window. For field descriptions, refer to the online help.
- **Step 7**: Click Save.

What to do next

Assign Authentication Proxy Rights, on page 272

Assign Authentication Proxy Rights

Perform the following procedure to enable authentication proxy rights for an existing user.

Procedure

- **Step 1**: Choose User Management > User Group. The Find and List User Group window appears.
- **Step 2**: Click Find.
- **Step 3**: Click the Standard EM Authentication Proxy Rights link. The User Group Configuration window appears.
- **Step 4**: Click Add End Users to Group. The Find and List Users window appears.
- **Step 5**: Click Find. You can also add a criteria for a specific user.
- **Step 6**: To assign proxy rights to one or more users, complete one of the following steps:
- **Step 7**: To add a single user, select the user and then click Add Selected.
- **Step 8**: To add all users that appear in the list, click Select All and then click Add Selected.
The user or users appear in the Users in Group table in the User Group Configuration window.

WebDialer Interactions and Restrictions

WebDialer Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Client Matter Codes (CMC)</td>
<td>When you use CMCs, you must enter the proper code at the tone; otherwise, the IP phone disconnects and the user receives a reorder tone.</td>
</tr>
<tr>
<td>Forced Authorization Codes (FAC)</td>
<td>When you use FACs, you must enter the proper code at the tone; otherwise, the IP phone disconnects and the user receives a reorder tone.</td>
</tr>
<tr>
<td>ApplicationDialRule table</td>
<td>Cisco WebDialer uses change notifications on the ApplicationDialRule database table to track and use updated dial rules.</td>
</tr>
<tr>
<td>Client Matter Codes and Forced Authorization Codes</td>
<td>Web Dialer supports CMCs and FACs in the following ways:</td>
</tr>
<tr>
<td></td>
<td>• A user can enter the destination number in the dial text box of the WD HTML page or SOAP request, and then manually enter the CMC or FAC on the phone.</td>
</tr>
<tr>
<td></td>
<td>• A user can enter the destination number followed by the FAC or CMC in the dial text box of the WD HTML page or SOAP request.</td>
</tr>
<tr>
<td></td>
<td>For example, if the destination number is 5555, the FAC is 111, and the CMC is 222, a user can make a call by dialing 5555111# (FAC), 5555222# (CMC), or 5555111222# (CMC and FAC).</td>
</tr>
<tr>
<td>Note</td>
<td>• WebDialer does not handle any validation for the destination number. The phone handles the required validation.</td>
</tr>
<tr>
<td></td>
<td>• If a user does not provide a code or provides the wrong code, the call will fail.</td>
</tr>
<tr>
<td></td>
<td>• If a user makes a call from the WebApp with a DN that contains special characters, the call goes successfully after stripping the special characters. The same rules do not work in SOAP UI.</td>
</tr>
</tbody>
</table>
WebDialer Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phones</td>
<td>Cisco WebDialer supports phones that run Skinny Client Control Protocol (SCCP) and Session Initiation Protocol (SIP) that Cisco Computer Telephony Integration (CTI) supports.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> Few older phone models do not support Cisco Web Dialer that run SIP.</td>
</tr>
</tbody>
</table>

WebDialer Troubleshooting

Authentication Error

**Problem**
Cisco WebDialer displays the following message:
Authentication failed, please try again.

**Possible Cause**
User entered wrong user ID or password.

**Solution**
Ensure that you use your Cisco Unified Communications Manager user ID and password to log in.

Service Temporarily Unavailable

**Problem**
Cisco WebDialer displays the following message:
Service temporarily unavailable, please try again later.

**Possible Cause**
The Cisco CallManager service became overloaded because it has reached its throttling limit of three concurrent CTI sessions.

**Solution**
After a short time, retry your connection.
Directory Service Down

Problem
Cisco WebDialer displays the following message:
Service temporarily unavailable, please try again later: Directory service down.

Possible Cause
The Cisco Communications Manager directory service may be down.

Solution
After a short time, retry your connection.

Cisco CTIManager Down

Problem
Cisco WebDialer displays the following message:
Service temporarily unavailable, please try again later: Cisco CTIManager down.

Possible Cause
Cisco CTIManager service that is configured for Cisco Web Dialer went down.

Solution
After a short time, retry your connection.

Session Expired, Please Login Again

Problem
Cisco WebDialer displays the following message:
Session expired, please login again.

Possible Cause
A Cisco Web Dialer session expires:
• After the WebDialer servlet gets configured
• If the Cisco Tomcat Service is restarted.

Solution
Log in by using your Cisco Unified Communications Manager User ID and Password.
User Not Logged In on Any Device

Problem
Cisco Web Dialer displays the following message:
User not logged in on any device.

Possible Cause
The user chooses to use Cisco Extension Mobility from the Cisco WebDialer preference window but does not get log in to any IP phone.

Solution
• Log in to a phone before using Cisco WebDialer.
• Choose a device from the Cisco WebDialer preference list in the dialog box instead of choosing the option Use Extension Mobility.

Failed to Open Device/Line

Problem
After a user attempts to make a call, Cisco WebDialer displays the following message:
User not logged in on any device.

Possible Cause
• The user chose a Cisco Unified IP Phone that is not registered with Cisco Unified Communications Manager. For example, the user chooses a Cisco IP SoftPhone as the preferred device before starting the application.
• The user who has a new phone chooses an old phone that is no longer in service.

Solution
Choose a phone that is in service and is registered with Cisco Unified Communications Manager.

Destination Not Reachable

Problem
Cisco WebDialer displays the following message on the End Call window:
Destination not reachable.

Possible Cause
• User dialed the wrong number.
• The correct dial rules did not get applied. For example, the user dials 5550100 instead of 95550100.
Solution

Check the dial rules.
Placing Calls

Destination Not Reachable
Paging Overview

Cisco Unified Communications Manager can be configured to integrate with Cisco Paging Server to provide basic paging services for Cisco Unified IP Phones and a variety of endpoints. The Cisco Paging Server product is offered through the InformaCast Virtual Appliance and offers the following deployment options:

InformaCast Basic Paging

InformaCast Basic Paging provides phone-to-phone live audio paging to individual Cisco IP phones or groups of up to 50 phones simultaneously. InformaCast Basic Paging is free to all Cisco Unified Communications Manager customers and all Cisco Business Edition 6000 and Cisco Business Edition 7000 customers.

InformaCast Advanced Notification

InformaCast Advanced Notification is a full-featured emergency notification and paging solution that allows you to reach an unlimited number of Cisco IP phones and various devices and systems with text and audio messages.

To streamline the configuration process, Cisco Unified Communications Manager comes with a provisioning wizard that allows you to quickly configure advanced notifications services.

Some of the features include:

- Text and audio (live or pre-recorded) to Cisco IP Phones and other endpoints
- Analog and IP overhead paging systems integration
- 911 or emergency call monitoring or alerting or recording
- Cisco Jabber integration
- Cisco Spark integration
• Automated weather notifications
• Dynamically triggered emergency conference calls
• Pre-recorded or scheduled broadcasts (school bells or shift changes)
• Event accountability with message confirmation and reporting
• Notification to computer desktops (Windows and Mac OS)
• Facilities integration (control lighting, door locks)
• Security integration (panic or duress buttons, motion detectors, fire)

Purchase a license key to access InformaCast Advanced Notification features.

InformaCast Mobile

InformaCast Mobile is a cloud-based service that allows users to send images, text, and pre-recorded audio to mobile devices running iOS or Android. It also has bi-directional integration with InformaCast Advanced Notification.

Some of the features include:
• The ability to send and receive InformaCast messages via mobile devices running iOS or Android
• Bi-directional integration with InformaCast Advanced Notification
• Message confirmations and read receipts
• No calling or SMS messaging fees

InformaCast Mobile must be purchased direct from Singlewire Software. Please refer to the Singlewire website for additional details and downloads.

If you have already configured Cisco Unified Communications Manager to integrate with InformaCast Advanced Notification, no further configuration of Cisco Unified Communications Manager is required.

Paging Prerequisites

Cisco Paging Server is designed to work in a multicast environment. You must configure your network for multicast.

For a list of Cisco Unified IP Phones that support paging, refer to the Cisco Unified IP Phones section of the Singlewire Compatibility Matrix at:

Cisco Unified Communications Manager Configuration for Basic Paging Task Flow

Perform the following tasks to configure Cisco Unified Communications Manager to integrate with Cisco Paging Server for an InformaCast Basic Paging deployment.

Before you begin

- Learn more about the feature by reviewing the following:
  - Paging Overview, on page 279
  - InformaCast Basic Paging, on page 279
- Review Paging Prerequisites, on page 280
- The configuration in this section is automated when using the Advanced Notification Paging Configuration Task Flow wizard.

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 SNMP Service, on page 282</td>
<td>Configure SNMP in Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Set Default Codec to G.711, on page 284</td>
<td>Set the default codec to G.711.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Device Pool for Paging, on page 284</td>
<td>Configure a device pool.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Route Partition for InformaCast Paging, on page 285</td>
<td>Configure a route partition for Basic Paging.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure Calling Search Space for InformaCast Paging, on page 286</td>
<td>Configure a calling search space for Basic Paging.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Configure CTI Ports for Paging, on page 286</td>
<td>Configure CTI ports.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Configure Access Control Group with AXL Access, on page 287</td>
<td>Configure an AXL access control group.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Configure Application User for Paging, on page 288</td>
<td>Configure an application user.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Enable web access for the phone using one of the following procedures:  - Enable Web Access for a Phone, on page 289  - Enable Web Access for Common Phone Profile, on page 289</td>
<td>You can enable web access on all phones globally using Enterprise Phone Configuration, a group of phones using a Common Phone Profile, or an individual phone.</td>
</tr>
</tbody>
</table>
Configure SNMP for Paging

Perform the following tasks to configure SNMP services in the cluster.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Enable SNMP Service, on page 282</td>
<td>Enable the SNMP and other services in the cluster.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Create an InformaCast SNMP Community String,</td>
<td>Configure an SNMP community string.</td>
</tr>
<tr>
<td>on page 283</td>
<td></td>
</tr>
</tbody>
</table>

**Enable SNMP Service**

To configure paging, you must enable SNMP on every node in the cluster. In addition, you must enable the following services:

- Cisco CallManager SNMP Service—Enable on all nodes in the cluster.
- Cisco CallManager—Enable on at least one node.
- Cisco AXL Web Services—Enable on at least one node.
- Cisco CTIManager—Enable on at least one node.

**Procedure**

**Step 1** Log in to Cisco Unified Serviceability and choose **Tools > Service Activation**.
**Step 2** From the **Server** drop-down list, choose the server on which you want to configure SNMP.
**Step 3** Check the check boxes that correspond to the **Cisco CallManager SNMP Service**.
**Step 4** For at least one server in the cluster, check the check boxes that correspond to **Cisco CallManager**, **Cisco CTIManager**, and **Cisco AXL Web Service** services.
Create an InformaCast SNMP Community String

Perform this procedure for Basic Paging to set up an SNMP community string.

**Before you begin**

Enable SNMP Service, on page 282

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified Serviceability, choose SNMP &gt; V1/V2c &gt; Community String.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list box, choose a server and click Find.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Add New.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Community String Name field, enter ICVA.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>From the Access Privileges drop-down menu, select ReadOnly.</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>Check the Apply to All Nodes check box if the check box is active.</td>
<td></td>
</tr>
<tr>
<td>Step 7</td>
<td>Click Save.</td>
<td></td>
</tr>
<tr>
<td>Step 8</td>
<td>Click OK.</td>
<td></td>
</tr>
</tbody>
</table>
Placing Calls

### Set Default Codec to G.711

You must create an InformaCast region that uses G.711 as the default codec for calls to other regions.

**Before you begin**

Configure SNMP for Paging, on page 282

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified CM Administration, choose System &gt; Region Information &gt; Region.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Name field, enter ICVA.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>In the Regions text box, select all regions by pressing the CTRL key and clicking all of the selected regions.</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Maximum Audio Bit Rate drop-down list box, select 64 kbps (G.722, G.711).</td>
<td></td>
</tr>
<tr>
<td>Step 7</td>
<td>From the Maximum Session Bit Rate for Video Calls column click the None radio button.</td>
<td></td>
</tr>
<tr>
<td>Step 8</td>
<td>Click Save.</td>
<td></td>
</tr>
</tbody>
</table>

**What to do next**

Configure a Device Pool for Paging, on page 284

### Configure a Device Pool for Paging

Perform this procedure to configure a device pool for your paging deployment.

**Before you begin**

Set Default Codec to G.711, on page 284

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified CM Administration, choose System &gt; Device Pool.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Device Pool Name field, enter ICVA.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Cisco Unified Communications Manager Group drop-down list box, select the group that contains the Cisco Unified Communications Manager cluster with which the InformaCast Virtual Appliance will communicate.</td>
<td></td>
</tr>
</tbody>
</table>
Step 5  From the Date/Time Group drop-down list box, select a date/time group. Select CMLocal unless you are performing dialing restrictions by the time of day.

Step 6  From the Region drop-down list box, choose ICVA.

Step 7  From the SRST Reference drop-down list box, select Disable.

Step 8  Click Save.

What to do next
Configure Partitions and Calling Search Spaces for Paging, on page 285

Configure Partitions and Calling Search Spaces for Paging

Perform the following tasks to configure a partition and calling search space (CSS) for paging as follows:
- For Basic Paging deployments, create a single partition and CSS for InformaCast paging.

Before you begin
Configure Region for Paging, on page 283

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1  Configure Route Partition for InformaCast Paging, on page 285</td>
<td>Configure a route partition for InformaCast paging.</td>
</tr>
<tr>
<td>Step 2  Configure Calling Search Space for InformaCast Paging, on page 286</td>
<td>Configure a calling search space for InformaCast paging.</td>
</tr>
</tbody>
</table>

Configure Route Partition for InformaCast Paging

Create a route partition for InformaCast paging.

Before you begin
Configure a Device Pool for Paging, on page 284

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose Call Routing &gt; Class of Control &gt; Route Partitions.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Name field, enter the following name and description for the partition: ICVA-CTIOutbound, ICVA-Do not add to any phone CSS.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>
Configure Calling Search Space for InformaCast Paging

Perform this procedure to configure a calling search space for InformaCast paging.

Before you begin

Configure Route Partition for InformaCast Paging, on page 285

Procedure

Step 1 In 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 ICVA.
Step 4 In the Available Partitions list box, use the arrows to move the following partitions to the Selected Partitions list box.
  • The partition that you created for InformaCast paging
  • The partitions that contain your users' extensions and any analog paging extensions
Step 5 Click Save.

What to do next

Configure CTI Ports for Paging, on page 286

Configure CTI Ports for Paging

Perform this procedure to configure CTI ports for your paging deployment. The number of CTI ports that you need depends on your deployment type and your applications' usage:

• For Basic Paging deployments, you must create a minimum of two CTI ports for InformaCast paging.

Before you begin

Configure Calling Search Space for InformaCast Paging, on page 286

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Add New.
Step 3 From the Phone Type drop-down list box, choose CTI Port.
Step 4 In the Device Name field, enter a name for the CTI Port. For example, ICVA-IC-001 for an InformaCast port.
Step 5  In the Description field, enter a description for the port. For example, InformaCast Recording Port for Call Monitoring.

Step 6  From the Device Pool drop-down list box, select ICVA.

Step 7  From the Calling Search Space drop-down list box, select ICVA.

Step 8  From the Device Security Profile drop-down list box, select Cisco CTI Port - Standard SCCP Non-Secure Profile.

Step 9  Click Save.

Step 10 Click OK.

Step 11 In the left association area, click Line [1] - Add a new DN.

Step 12 In the Directory Number field, enter a directory number. This directory number should not be used for any purpose other than making paging calls. It should not be assigned to a phone and should not be within a direct-inward-dialing range.

Step 13 In the Route Partition drop-down list box, select the following ports:
  * For InformaCast ports, select ICVA-CTIOutbound.

Step 14 In the Display (Internal Caller ID) text box, enter InformaCast.

Step 15 In the ASCII Display (Internal Caller ID) text box, enter InformaCast.

Step 16 Click Save.

Step 17 Repeat this procedure for each CTI port that you need.

---

**What to do next**

Configure Access Control Group with AXL Access, on page 287

---

**Configure Access Control Group with AXL Access**

Perform this procedure to create an access control group that includes AXL access.

**Before you begin**

Configure CTI Ports for Paging, on page 286

**Procedure**

---

Step 1  In Cisco Unified CM Administration, choose User Management > User Settings > Access Control Group.

Step 2  Click Add New.

Step 3  In the Name text box, enter ICVA User Group.

Step 4  Click Save.

Step 5  From the Related Links drop-down, select Back to Find/List and click Go.

Step 6  In the Roles column, click the i icon that corresponds to the new access control group.

Step 7  Click Assign Role to Group.

Step 8  Click Find.

Step 9  Select Standard AXL API Access check box, and click Add Selected.
Configure Application User for Paging

Perform this procedure to configure an application user:

• For Basic Paging, configure an InformaCast application user.

Before you begin
Configure Access Control Group with AXL Access, on page 287

Procedure

Step 1 In Cisco Unified CM Administration, choose User Management > Application User.
Step 2 Click Add New.
Step 3 In the UserID text box, enter a userID for the application user. For example, ICVA InformaCast.
Step 4 Enter a password in the Password and Confirm Password fields.
Step 5 In the Available Devices list box, click the CTI ports that you created for your deployment and use the arrows to move the devices to the Controlled Devices list box. For example, select ICVA-IC-001 for InformaCast and ICVA-CA-001 for CallAware.
Step 6 Click the Add to Access Control Group button.
Step 7 Click Find.
Step 8 Check the following check boxes (unless otherwise indicated, select these permissions for all application users):

• ICVA User Group
• Standard CTI Allow Control of All Devices
• Standard CTI Allow Control of Phones supporting Connected Xfer and conf
• Standard CTI Allow Control of Phones supporting Rollover Mode
• Standard CTI Enabled

Step 9 Click Add Selected.
Step 10 Click Save.

What to do next
Perform one of the following procedures to enable web access on your phones:

• Enable Web Access for a Phone, on page 289
• Enable Web Access for Common Phone Profile, on page 289
Enable Web Access for a Phone

Perform this procedure in Basic Paging to enable web access for a Cisco Unified IP Phone. You can also use a Common Phone Profile to enable web access for a group of phones that use that profile. For details, see Enable Web Access for Common Phone Profile, on page 289.

Before you begin
Configure Application User for Paging, on page 288

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 Device &gt; Phone.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select the phone for which you want to enable web access.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Product Specific Configuration Layout area, from the Web Access drop-down menu, select Enabled.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

What to do next
Configure Authentication URL, on page 290

Enable Web Access for Common Phone Profile

Perform this procedure in Basic Paging to enable web access for a group of Cisco Unified IP Phones that use a Common Phone Profile. You can also enable web access on an individual phone. For details, see Enable Web Access for a Phone, on page 289.

Before you begin
Configure Application User for Paging, on page 288

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 Device &gt; Device Settings &gt; Common Phone Profile.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find and select the profile that applies to the group of phones for which you want to enable web access.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Product Specific Configuration Layout area, from the Web Access drop-down list, select Enable.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Apply Config to reset the phones that use the Common Phone Profile.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click OK.</td>
</tr>
</tbody>
</table>
Enable Web Access for Enterprise Phone Configuration

Perform this procedure in Cisco Unified Communications Manager to enable web access for a group of Cisco Unified IP Phones that use a Common Phone Profile. You can also enable web access on an individual phone. For more details, see Enable Web Access for a Phone, on page 289.

Before you begin
Configure Application User for Paging, on page 288.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose System &gt; Enterprise Phone Configuration.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>From the Web Access drop-down list, select Enable.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Apply Config to reset the phones that use the Common Phone Profile.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click OK.</td>
</tr>
</tbody>
</table>

What to do next
Configure Authentication URL, on page 290.

Configure Authentication URL

Perform the following tasks to configure an authentication URL that points to InformaCast so that when InformaCast pushes broadcasts to Cisco Unified IP Phones, the phones authenticate with InformaCast instead of Cisco Unified Communications Manager.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1  Set Authentication URL, on page 291</td>
<td>Set the Cisco Unified Communications Manager authentication URL to point InformaCast.</td>
</tr>
<tr>
<td>Step 2  Reset Your Phones, on page 291</td>
<td>Reset the phones in your deployment so that your phones use the new settings.</td>
</tr>
<tr>
<td>Step 3  Test Your Phones, on page 292</td>
<td>Verify that the phones in your deployment use the new authentication URL settings.</td>
</tr>
</tbody>
</table>
Set Authentication URL

Perform this procedure to set the Cisco Unified Communications Manager authentication URL to point to the InformaCast Virtual Appliance.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>In Cisco Unified CM Administration choose <strong>System &gt; Enterprise Parameters</strong>.</td>
</tr>
<tr>
<td>2</td>
<td>Scroll to the <strong>Phone URL Parameters</strong> area, and in the <strong>URL Authentication</strong> field, enter <code>http://&lt; IPAddress &gt;:8081/InformaCast/phone/auth</code> where <code>&lt; IPAddress &gt;</code> is the IP Address of the InformaCast Virtual Appliance.</td>
</tr>
<tr>
<td>3</td>
<td>Make a note of the existing URL in the <strong>URL Authentication</strong> field. You may need this when you configure InformaCast. See your InformaCast documentation for details.</td>
</tr>
<tr>
<td>4</td>
<td>Scroll to the <strong>Secured Phone URL Parameters</strong> area, and in the <strong>Secured Authentication URL</strong> field, enter <code>http://&lt; IPAddress &gt;:8081/InformaCast/phone/auth</code> where <code>&lt; IPAddress &gt;</code> is the IP Address of the InformaCast Virtual Appliance.</td>
</tr>
<tr>
<td>5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**What to do next**

[Reset Your Phones, on page 291](#).

Reset Your Phones

After you set the authentication URL to point to the InformaCast Virtual Appliance, you must reset your phones. This procedures describes how to manually reset the phones in device pools. There are many methods for resetting your phones. For example, you can also use Bulk Administration Tool to schedule the reset during off hours. See the Cisco Unified Communications Manager Bulk Administration Guide for information on the Bulk Administration Tool.

**Before you begin**

[Set Authentication URL, on page 291](#).

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>In Cisco Unified CM Administration, choose <strong>System &gt; Device Pool</strong>.</td>
</tr>
<tr>
<td>2</td>
<td>Click <strong>Find</strong>.</td>
</tr>
<tr>
<td>3</td>
<td>Click the name of the Device Pool that you want to reset.</td>
</tr>
<tr>
<td>4</td>
<td>Click <strong>Reset</strong>. The <strong>Device Reset</strong> dialog box is displayed.</td>
</tr>
<tr>
<td>5</td>
<td>Click <strong>Reset</strong>.</td>
</tr>
</tbody>
</table>
Test Your Phones

Verify that your phones are authenticating with the InformaCast Virtual Appliance.

Before you begin

Reset Your Phones, on page 291.

Procedure

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

Step 2
Use the drop-down menus and fields in the Find and List Phones window to filter your search for a phone that should be using the new authentication URL, and click Find.

Step 3
For the phone that should be using the new settings, click the IP Address link in the IPv4 Address column.

Step 4
Click Network Configuration.
The Network Configuration page appears.

Step 5
Verify that the Authentication URL field displays the InformaCast Virtual Appliance IP address that you entered for the URL Authentication enterprise parameter. If the correct URL does not appear, you will need to set the authentication URL.

What to do next

For Basic Paging, you can now start using paging services.

Advanced Notification Paging Configuration Task Flow

Perform the following tasks to integrate InformaCast Paging Server with Cisco Unified Communications Manager for IP paging and emergency call alerting. It includes the following features:

- InformaCast advanced notification
- Panic button configuration
- Text and audio notification to IP phones when a user dials an emergency services number (CallAware)

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | Install the InformaCast Virtual Appliance, on page 293.  
Download the InformaCast OVA file from the Singlewire website and upload it to vSphere. |
| **Step 2** | Configure Connection to InformaCast, on page 295.  
Configure Cisco Unified Communications Manager and InformaCast. |
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Panic Button, on page 296. Configure a panic button to send a text and audio notification to IP phones.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure CallAware Emergency Call Alerting, on page 298. Configure emergency call text and audio notifications.</td>
</tr>
</tbody>
</table>

## Install the InformaCast Virtual Appliance

Singlewire supports InformaCast Virtual Appliance on the VMware ESXi platform, which is managed through the vSphere client.

---

**Note**

To view a list of Singlewire-supported VMware ESXi versions, go to this URL: [https://www.singlewire.com/compatibility-matrix](https://www.singlewire.com/compatibility-matrix) and click the Server Platforms link under InformaCast Platform section.

---

**Note**

If you have purchased a license, refer to [https://www.singlewire.com/icva-kb-activate](https://www.singlewire.com/icva-kb-activate) to activate your license. This will ensure that Emergency Notifications stay active after the 90-day trial.

---

**Note**

For more details on the installation, including InformaCast screen captures, go to this URL: [https://www.singlewire.com/icva-kb-install](https://www.singlewire.com/icva-kb-install).

---

**Before you begin**

Import InformaCast Virtual Appliance using the vSphere client. This can be downloaded from your VMware server.

**Procedure**

---

**Step 1**

Download the OVA file from the Singlewire website and then log in to the vSphere client.

**Note**

If you are using InformaCast on the Communications Manager Business Edition 6000, you are supplied with a DVD in a package with an OVA on it (physical media).

The vSphere Client window appears.

**Step 2**

From the vSphere Client window, choose File > Deploy OVF Template. The Deploy OVF Template dialog box appears.

**Step 3**

Click the Deploy from File radio button and then click Browse to select the saved the OVA file (or to the OVA file on the supplied DVD). After you select the OVA file, click Open. The Source location is selected in the Deploy OVF Template dialog box.

**Step 4**

Click Next to continue. The Deploy OVF Template dialog box refreshes and OVF Template Details appears.
Install the InformaCast Virtual Appliance

Step 5 Click **Next** to verify the **Name and Location**, and then click **Next** to select the network to store the new virtual machine files.

**Tip** It is good practice to place the Virtual Appliance on the same VLAN as your Cisco Unified Communications Manager.

Step 6 Click **Next** to continue, and then click **Finish**. The InformaCast Virtual Appliance begins importing.

Step 7 From the **vSphere Client** window, click **Hosts and Clusters** icon and then select your host server. The **vSphere Client** window refreshes.

Step 8 Click the **Configuration** tab and select the **Virtual Machine Startup/Shutdown** link in the **Software** section.

Step 9 Click the **Properties** link. The **Virtual Machine Startup and Shutdown** dialog box appears.

Step 10 Check the **Allow virtual machines to start and stop automatically with the system** check box under **System Settings**.

Step 11 Under **Startup Order**, scroll to the **Manual Startup** section and select your virtual machine (by default, this is Singlewire InformaCast VM), and then move it from the **Manual Startup** section to the **Automatic Startup** section, by using the **Move Up** button. After moving it, click **OK**. The InformaCast Virtual Appliance starts and stops automatically with the server on which it is hosted. Now you can turn on InformaCast’s virtual machine and set its network configuration.

Step 12 Choose **View** > **Inventory** > **VMs and Templates** and then select your virtual machine.

Step 13 Choose the **Inventory** > **Virtual Machine** > **Open Console**. The Singlewire InformaCast VM console window appears.

Step 14 InformaCast configuration starts for the first time. During this configuration, perform the following tasks for the InformaCast Virtual Appliance:

a) Accept Cisco End User License Agreement (EULA)
b) Accept Singlewire EULA
c) Set up hostname
d) Set up IP address, subnet mask, and default gateway
e) Set up DNS server IP address and domain name
f) Set up NTP server IP address or hostname
g) Set up time zone
h) Set up Secure Socket Layer (SSL) certificate parameters
i) Set up SSL subject alternate names (optional)
j) Set up the OS admin password
k) Set up the InformaCast and PTT (PushToTalk) admin password. This password is required to connect the Cisco Unified Communications Manager and InformaCast in the Cisco Unified CM Administration, 

   **Advanced Features** > **Emergency Notifications Paging**.

l) Set up security passphrase for backup and communication

When your configuration is successful, the “Welcome to Singlewire InformaCast” message is displayed.

Step 15 Click **Continue** to work with Singlewire InformaCast.

**What to do next**

Configure Connection to InformaCast, on page 295.
Configure Connection to InformaCast

Use this procedure to load the InformaCast certificate to the Cisco Unified Communications Manager Tomcat trust store.

Before you begin

Install the InformaCast Virtual Appliance, on page 293.

Procedure

**Step 1**
From Cisco Unified CM Administration, choose **Advanced Features > Emergency Notifications Paging.** The **Emergency Notifications Paging** wizard appears.

**Step 2**
In the **Introduction to InformaCast Emergency Notifications** page, click **Next** to continue. The **Installing the InformaCast Virtual Appliance** page appears.

**Step 3**
In the **Installing the InformaCast Virtual Appliance** page, click **Next** to continue.

**Note** You should have successfully installed InformaCast Virtual Appliance to configure with the Cisco Unified Communications Manager.

The **Connecting Cisco Unified Communications Manager and InformaCast** page appears.

**Step 4**
In the **IP address of InformaCast VM** field, enter either IP address or Hostname.

**Note** By default, the username is stated as **admin** in the **Username to use in InformaCast** field, and it is not editable.

**Step 5**
In the **Password for admin app user** field, enter the administrator password of the InformaCast application. The dialog box displaying the thumbprint of InformaCast certificate is displayed.

**Step 6**
Click **OK** to load the InformaCast certificate to the Cisco Unified Communications Manager Tomcat trust store.

Configuration process starts.

**Note** When the configuration is successful, the **Status** field displays the completion status.

**Step 7**
Click **Next**. The wizard performs the following tasks:

- Activates SNMP service
- Configures SNMP Service with locally generated random credentials
- Activates CTI Manager Service
- Configures Cisco Unified Communications Manager for InformaCast
  - Creates new region (1 per cluster)
  - Creates new device pool (1 per cluster)
  - Creates SIP trunk (1 per cluster)
  - Creates route group (1 per cluster)
  - Creates route list
• Creates role
• Creates app user

• Configures InformaCast for Cisco Unified Communications Manager
  • Creates a cluster
  • Refreshes recipient groups
  • Sets SIP access to deny
  • Creates SIP access

---

**What to do next**

Configure Panic Button, on page 296.

---

### Configure Panic Button

Use this procedure to configure a panic button to send a text and audio notification to IP phones. This allows you to initiate a one click alarm if there is emergency.

**Before you begin**

Configure Connection to InformaCast, on page 295.

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Advanced Features > Emergency Notifications Paging. The Emergency Notifications Paging wizard appears.

**Step 2** In the Introduction to InformaCast Emergency Notifications page, click Next to continue.

**Step 3** In the Installing the InformaCast Virtual Appliance page, click Next to continue.

**Step 4** In the Connecting Cisco Unified Communications Manager and InformaCast page, click Next to continue. The Configuring a Panic Button page appears.

**Step 5** From the Choose pre-recorded message by name drop-down list, select the pre-recorded message to be displayed on Cisco Unified IP phones and various devices and systems in emergency.

**Note** You can change the pre-recorded message in InformaCast administration, as required.

**Step 6** In the Enter DN to trigger the panic button field, enter the Directory Number (DN), which includes the digits 0 to 9, asterisks (*), and pound signs (#). Default value is **5.

**Step 7** From the Route Partition drop-down list box, select a partition to restrict access to the route pattern.

**Note** If you do not want to restrict access to the route pattern, select <None> for the partition.

**Step 8** Click Choose Phones to Send Notification button. The Phones to Send Notification dialog box appears.
Step 9  From the **Phones to Send Notification** dialog box, select the Cisco Unified IP phones to send the pre-recorded message. The dial pattern entered by you (for example, ***5) is configured as speed dial on the selected phones.

The selected Cisco Unified IP phones are displayed in the **Selected Phones to Send Notification** list box.

Step 10  Click **Add Rules**, to create a new rule for the selected Cisco Unified IP Phones to receive notifications.

a) Select one of the parameters from the drop-down list box. The available options are Device Pool, Description, and Directory Number.

b) In the second drop-down list box, select a criteria from the following options:

- Does
- Does not

c) In the third drop-down list box, select a criteria from the following options:

- Begins with
- Ends with
- Contains

d) In the text box, enter the search criterion.

**Note**  Minimum of one new rule and maximum of new five rules can be created. The **Add Rules** button gets disabled when five rules are created.

**Note**  To delete a rule, click **Delete Rules** button.

e) Click **Test Rules**, to validate the created rules. When the test rule is completed with more than zero phones, the **Next** button is enabled.

**Note**  Phones added to Cisco Unified Communications Manager at a later date that match this rule will be included as recipients in notifications to this group.

Step 11  Click **Next**.

The wizard performs the following tasks:

- Adds a speed dial for the entered DN to the selected phones. If the selected phones have unused speed dials assigned to existing phone button templates, this speed dial appears directly on the selected phones. If the selected phones do not have unused speed dial buttons, the panic button speed dial is created, but it does not appear on the phone.

- Adds route pattern for entered DN in selected partition using created route list.

- Creates an InformaCast DialCast entry for the entered DN to send the selected message to the phones matching the selected rules.

---

**What to do next**

Configure CallAware Emergency Call Alerting, on page 298.
Configure CallAware Emergency Call Alerting

Use this procedure to configure the CallAware emergency call alerting details. This sends a text and audio notification to IP phones when an emergency number is dialed. It can also detect calls to numbers other than 911.

**Before you begin**
Configure Panic Button, on page 296.

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose Advanced Features > Emergency Notifications Paging.  
The Emergency Notifications Paging wizard appears.

**Step 2**  
In the Introduction to InformaCast Emergency Notifications page, click Next to continue.

**Step 3**  
In the Installing the InformaCast Virtual Appliance page, click Next to continue.

**Step 4**  
In the Connecting Cisco Unified Communications Manager and InformaCast page, click Next to continue.

**Step 5**  
In the Configuring a Panic Button page, click Next to continue.  
The Configuring CallAware Emergency Call Alerting page appears.

**Step 6**  
From the Choose pre-recorded message by name drop-down list, select the pre-recorded message to be displayed on Cisco Unified IP phones and various devices and systems in emergency.

**Note**  
You can change the pre-recorded message in InformaCast administration, as required.

**Step 7**  
Click Choose Emergency Route Patterns button.  
The Route Patterns dialog box appears.

**Step 8**  
From the Route Patterns dialog box, select the route patterns by checking the box next to the desired patterns.  
a) Click the Save Selected/Changes button.

The selected route patterns are displayed in the Selected Route Patterns list box.

**Step 9**  
Click Add Rules, to create a new rule for the selected Cisco Unified IP Phones to receive notifications.

a) Select one of the parameters from the drop-down list box. The available options are Device Pool, Description, and Directory Number.

b) In the second drop-down list box, select a criteria from the following options:

- Does
- Does not

c) In the third drop-down list box, select a criteria from the following options:

- Begins with
- Ends with
- Contains

d) In the text box, enter the search criterion.

**Note**  
Minimum of one new rule and maximum of five new rules can be created. The Add Rules button gets disabled when five rules are created.

**Note**  
To delete a rule, click Delete Rules button.
e) Click Test Rules, to validate the created rules. When the test rule is completed with more than zero phones, the Finish button is enabled.

**Note**  Phones added to Cisco Unified Communications Manager at a later date that match this rule will be included as recipients in notifications to this group.

**Step 10**

Click Finish.

The wizard performs the following tasks:

- Adds External Call Control profile for InformaCast
- For each selected route pattern, modify that route pattern to reference the External Call Control profile
- Creates a recipient group with rules that match phones to receive the notification
- Creates an InformaCast routing request with the selected message and recipient group

The Summary page appears and confirms the successful configuration of InformaCast with Cisco Unified Communications Manager. For more information, see [https://www.singlewire.com](https://www.singlewire.com).

---

**Paging Interactions**

- Advanced Notification Paging Interactions, on page 299

**Advanced Notification Paging Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Emergency Notifications Paging</td>
<td>You can configure the Emergency Notifications Paging wizard using InformaCast Release 11.5(1)SU3 and later versions in basic paging mode only.</td>
</tr>
<tr>
<td></td>
<td>You can configure call monitoring to route patterns that contain digits only in the Emergency Notifications Paging wizard. For route patterns that contain wildcard characters, configure in InformaCast.</td>
</tr>
</tbody>
</table>
Intercom

Intercom Overview

Intercom is a type of phone line that combines the functionality of a traditional line and a speed dial. With an intercom line, a user can call the intercom line of another user, which answers automatically to one-way audio whisper. The recipient can then acknowledge the whispered call and initiate a two-way intercom call.

You can use an intercom line to dial any other intercom line in the intercom partition, or you can preconfigure the line to target an intercom line outside the intercom partition.

Intercom allows a user to place a call to a predefined target. The called destination answers the call automatically in speakerphone mode with mute activated. This sets up a one-way voice path between the initiator and the destination, so the initiator can deliver a short message, regardless of whether the called party is busy or idle.

To ensure that the voice of the called party does not get sent back to the caller when the intercom call is automatically answered, Cisco Unified Communications Manager implements whisper intercom. Whisper intercom ensures that only one-way audio exists from the caller to the called party. The called party must manually press a key to talk to the caller.

An auto-answer tone indicates the beginning of the whisper intercom state for both the sender and the recipient.

Intercom and Default Devices

Each intercom line needs a default device. The intercom line is displayed only on the designated default device.

When the administrator assigns an intercom line to a device, the system sets the device as the default device for the intercom line if not set previously. The administrator can modify the default device for the intercom line. When the administrator changes the default device to a different device, the intercom line gets removed from the original device, even though the intercom line may still be assigned to the original device.
You can assign an intercom line to a device profile. Only when a user uses a device profile to log in to the default device that matches the default device of the intercom line does the intercom line become available. Otherwise, no intercom line is displayed when the user logs in.

## Intercom Prerequisites

The intercom feature has the following system requirements:

- Cisco Unified IP Phones Firmware Release 8.3(1) or later

## Intercom Configuration Task Flow

**Before you begin**

- Review Intercom Prerequisites, on page 302.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
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<td>Configure Intercom Partition, on page 302</td>
<td>To add a new Intercom partition or configure an existing partition.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure an Intercom Calling Search Space, on page 303</td>
<td>To add a new Intercom Calling Search Space.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure an Intercom Translation Pattern, on page 304</td>
<td>To add a new Intercom Translation Pattern or to configure an existing Intercom Translation Pattern.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure an Intercom Directory Number, on page 304</td>
<td>To add or update an Intercom Directory Number.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Intercom Line and Speed Dial Configuration, on page 305</td>
<td>Configure Intercom Line and Speed Dial.</td>
</tr>
</tbody>
</table>

## Configure Intercom Partition

**Before you begin**

Ensure the phone model supports the Intercom feature for a particular release and device pack Generate a Phone Feature List, on page 1

**Procedure**

**Step 1**

In the Cisco Unified Communications Manager Administration window, choose Call Routing > Intercom > Intercom Route Partition.
The Find and List Intercom Partitions window appears.

**Step 2**
Click Add New.

An Add New Intercom Partition window appears.

**Step 3**
Under the Intercom Partition Information section, in the Name box, enter the name and description of the intercom partition that you want to add.

**Note**
To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (,) to separate the partition name and description on each line. If a description is not entered, Unified Communications Manager uses the partition name as the description.

**Step 4**
Click Save.

**Step 5**
Locate the partition that you want to configure.

Intercom Partition Configuration window is displayed

**Step 6**
Configure the fields in the Intercom Partition Configuration field area. See the online help for more information about the fields and their configuration options.

**Step 7**
Click Save.

The Intercom Partition Configuration window appears.

**Step 8**
Enter the appropriate settings. For detailed information about the Intercom Partition Configuration parameters, see online help.

**Step 9**
Click Save.

**Step 10**
Click Apply Config.

---

**What to do next**
Configure an Intercom Calling Search Space, on page 303

### Configure an Intercom Calling Search Space

**Before you begin**
Configure Intercom Partition, on page 302

**Procedure**

**Step 1**
In the menu bar, choose Call Routing > Intercom > Intercom Calling Search Space.

**Step 2**
Click the Add New button.

**Step 3**
Configure the fields in the Intercom Calling Search Space field area. See the online help for more information about the fields and their configuration options.

**Step 4**
Click Save.
Configure an Intercom Translation Pattern

Before you begin
Configure an Intercom Calling Search Space, on page 303

Procedure

Step 1  Choose Call Routing > Intercom > Intercom Translation Pattern. The Find and List Intercom Translation Patterns window appears.

Step 2  Perform one of the followings tasks:
    a) To copy an existing intercom translation pattern, locate the partition to configure, click Copy eside the intercom translation pattern to copy.
    b) To add a new intercom translation pattern, click the Add New button.

Step 3  Configure the fields in the Intercom Translation Pattern Configuration field area. For more information on the fields and their configuration options, see the system Online Help.

Step 4  Click Save.

Ensure that the intercom translation pattern that uses the selected partition, route filter, and numbering plan combination 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.

The Intercom Translation Pattern Configuration window displays the newly configured intercom translation pattern.

What to do next
Configure an Intercom Directory Number, on page 304

Configure an Intercom Directory Number

You can assign patterns to intercom directory numbers; for example, 352XX. To avoid user confusion, when you assign a pattern to an intercom directory number, add text or digits to these intercom DN configuration fields, Line Text Label, Display (Internal Caller ID), and External Phone Number Mask. These fields are displayed for an intercom directory number only after you add the intercom directory number and you associate the intercom directory number with a phone.

For example, add the username to the line text label and internal caller ID, and add the outside line number to the external number mask, when the calling information is displayed, it says John Chan, not 352XX.
Procedure

**Step 1** Choose Call Routing > Intercom > Intercom Directory Number.

The Find and List Intercom Directory Numbers window is displayed.

**Step 2** To locate a specific intercom directory number, enter search criteria and click Find.

A list of intercom directory numbers that match the search criteria displayed.

**Step 3** Perform one of the followings tasks:

a) To add an intercom directory number, click Add New.

b) To update an intercom directory number, click the intercom directory number to update.

The Intercom Directory Number Configuration window displayed.

**Step 4** Configure the fields in the Intercom Directory Number Configuration field area. For more information on the fields and their configuration options, see the system Online Help.

**Step 5** Click Save.

**Step 6** Click Apply Config.

**Step 7** Click Reset Phone.

**Step 8** Restart devices.

During the restart, the system may drop calls on gateways.

---

**Intercom Line and Speed Dial Configuration**

**Before you begin**

Configure an Intercom Directory Number, on page 304

**Procedure**

**Step 1** Choose Device > Device Settings > Phone Button Template and add the intercom line to an existing phone button template or create a new template.

**Note** The intercom line cannot be configured as the primary line.

**Step 2** From the Button Information area, from Feature drop-down list, choose Intercom.

**Step 3** From the Button Information area, from Feature drop-down list, choose Speed Dial.

**Note** You can configure the intercom line with a predefined destination (speed dial) to allow fast access.

**Step 4** Click Save.

**Step 5** Click Apply Config.
## Intercom Interactions and Restrictions

### Intercom Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bulk Administration Tool</td>
<td>The Cisco Unified Communications Manager administrator can use the Bulk Administration Tool to add many intercom users at once instead of adding users individually. See the Cisco Unified Communications Manager Bulk Administration Guide for more information.</td>
</tr>
<tr>
<td>Barge</td>
<td>When the intercom destination is a barge target, the Cisco Unified IP Phone can still support whisper intercom.</td>
</tr>
<tr>
<td></td>
<td>When the destination caller chooses to talk to the intercom caller by pressing the intercom button, the original call is put on hold, and the barge initiator is released.</td>
</tr>
<tr>
<td>Do Not Disturb (DND)</td>
<td>The intercom call will override DND on the destination phone.</td>
</tr>
<tr>
<td>Call Preservation</td>
<td>When a call is preserved, the end user must hang up before the phone can reregister with Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td></td>
<td>When the intercom call is in whisper mode, it represents a one-way medium, and the terminating side might have no user at all; therefore, only the intercom call in talkback mode will get preserved. (Whisper intercom will not get preserved.)</td>
</tr>
<tr>
<td>Cisco Unified Survivable Remote Site Telephony (SRST)</td>
<td>When Cisco Unified IP Phones register with SRST, the phones do not register intercom lines; therefore, the feature will not be available when the phones are registered with SRST.</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Assistant</td>
<td>With the Cisco Unified Communications Manager Assistant Configuration Wizard, Cisco Unified Communications Manager Assistant configuration takes less time and eliminates errors. The partitions, calling search spaces, route point, and translation pattern automatically get created when the administrator successfully runs and completes the configuration wizard.</td>
</tr>
<tr>
<td>CTI</td>
<td>You can use CTI/JTAPI/TSP to set or modify the preconfigured target directory number for an intercom line. You will receive notification if the target directory number is updated or reconfigured through Cisco Unified Communications Manager Administration.</td>
</tr>
<tr>
<td></td>
<td>Be aware that CTI/JTAPI/TSP is backward compatible if the intercom line is not configured to be controlled by the application. If the intercom line is configured in the application user list, you may have to make changes and test the compatibility.</td>
</tr>
<tr>
<td>Cisco Extension Mobility</td>
<td>The intercom feature interacts with Cisco Extension Mobility. The system presents an intercom line to a user who uses Cisco Extension Mobility to log in to a phone that supports the feature if the device profile that the user uses to log in has an intercom line that is provisioned. The phone must be the default device for that intercom line.</td>
</tr>
</tbody>
</table>
Interaction Feature

Internet Protocol Version 6 (IPv6)

Intercom can support phones with an IP Addressing Mode of IPv4 Only or IPv4 and IPv6. During an intercom call, the talkback mode establishes media streams with the same IP version as the media stream that is used when the caller initiates intercom.

Intercom directory numbers (lines)

Intercom directory numbers (lines) are restricted to one device per intercom line. Cisco Extension Mobility is widely used; mobile users need the intercom feature but need it to be available only on a single device. You can assign intercom lines to either a regular device or to an extension mobility profile, but the system needs to enforce that an intercom line gets associated to either a regular device or to an extension mobility profile.

Extension mobility profile

An extension mobility profile can be used on more than one phone simultaneously, use the Default Activated Device field in the Intercom Directory Number Configuration window (Cisco Unified CM Administration > Call Routing > Intercom > Intercom Directory Number Configuration) to specify which device can display this intercom line. Intercom lines that are not used for Extension Mobility also require configuration of the Default Activated Device field.

Intercom Restrictions

The following restrictions apply to the Intercom feature:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold</td>
<td>The system does not allow intercom calls to be placed on hold.</td>
</tr>
<tr>
<td>Call Forwarding</td>
<td>Intercom calls cannot be forwarded.</td>
</tr>
<tr>
<td>Transfer</td>
<td>The system does not allow an intercom call to be transferred.</td>
</tr>
<tr>
<td>iDivert</td>
<td>The system does not allow an intercom call to be diverted.</td>
</tr>
<tr>
<td>Call Pickup/Directed Call</td>
<td>The call pickup groups do not include intercom calls.</td>
</tr>
<tr>
<td>Pickup</td>
<td></td>
</tr>
<tr>
<td>DND</td>
<td>Intercom overrides Do Not Disturb (DND).</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>If sufficient bandwidth does not exist, the intercom call fails.</td>
</tr>
<tr>
<td>Call Target</td>
<td>If two intercom calls are directed to a target, the first one goes through; the second fails with a busy tone.</td>
</tr>
<tr>
<td>Barge and cBarge</td>
<td>Intercom does not work with Barge and cBarge.</td>
</tr>
<tr>
<td>Conferencing</td>
<td>The system does not allow intercom calls to be in conference.</td>
</tr>
<tr>
<td>Monitoring and Recording</td>
<td>When an active call is being monitored or recorded, the user cannot receive nor place intercom calls.</td>
</tr>
<tr>
<td>Video</td>
<td>Video is not supported with intercom.</td>
</tr>
</tbody>
</table>
### Intercom Troubleshooting

#### Busy Tone When Dialing Out of Intercom Line

**Problem**
Phone plays busy tone when user is dialing out of intercom line.

**Possible Cause**
The DN is not in the same intercom partition as the calling number.

**Solution**
- Ensure that the DN is in the same intercom partition as the calling number.
- If it is, ensure that the dialed-out DN is configured on another phone and that the phone is registered with the same Cisco Unified Communications Manager cluster.

#### Intercom Calls cannot use Talkback with Speaker, Handset or Headset

**Problem**
User cannot go into talkback mode for intercom calls by using headset, handset, or speaker.

**Possible Cause**
This situation exists by design. The only way to go into the connected state for intercom calls is by pressing the corresponding line button.

**Solution**
User can end call by using speaker, handset, or headset.

---

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom Partition</td>
<td>An intercom partition assigned to an item such as calling search space or to a route pattern cannot be deleted.</td>
</tr>
<tr>
<td>Intercom Calling Search Spaces</td>
<td>Intercom calling search spaces that devices, lines (DNs), translation patterns, or other items are using cannot be deleted.</td>
</tr>
</tbody>
</table>
Troubleshooting SCCP

Intercom Lines Not Showing Up on Phone

Problem
Intercom lines do not display on the phone.

Possible Cause
The phone version may be earlier than 8.3(1), or the button template may not be assigned to the phone.

Solution
• Check the phone version. Ensure that it is 8.3(1) or later.
• Determine whether the button template is assigned to the phone.
• Capture the sniffer trace between Cisco Unified Communications Manager and the phone. In the button template response, see whether intercom lines get sent to the phone (button definition = Ox17).

Intercom Lines Not Showing Up When Phone Falls Back to SRST

Problem
A phone that was configured with Cisco Unified Communications Manager Release 6.0(x) or later, includes two intercom lines. Cisco Unified Communications Manager stops and falls back to SRST. The intercom lines do not display.

Possible Cause
The SCCP version of SRST does not support SCCP Version 12.

Solution
• Check the SCCP Version of SRST. If SRST supports SCCP Version 12, it will support intercom lines.
• If SRST supports SCCP Version 12, capture a sniffer trace and ensure that the button template that the phone sent includes intercom lines.

Troubleshooting SIP

Debug Phones That Are Running SIP

Use this debug command: Debug sip-messages sip-task gsmsmslm sip-adapter.

Configuration of Phones That Are Running SIP

Show config — The command on the phone is displayed if intercom lines are configured as regular lines with featureid-->23.
Cisco Extension Mobility User Is Logged In But Intercom Line Does Not Display

Problem
The Cisco Extension Mobility user is logged in to a phone, but the user intercom line does not display.

Possible Cause
Default Activated Device is configured incorrectly.

Solution
• Check that the Default Activated Device is configured on the intercom directory number.
• Check that the Default Activated Device matches the device to which the user is logged in.

Intercom Line Fails to Display on Phone

Problem
An intercom line has been configured and assigned to a phone but fails to display on the phone.

Possible Cause
Default Activated Device value is set to the intercom line of this device.

Solution
If the configuration has been done, reset the phone.
PART IX

Receiving Calls

• Prime Line Support, on page 313
• Call Forwarding, on page 317
• Call Pickup, on page 343
• Call Park and Directed Call, on page 365
• Extension Mobility, on page 393
• Extension Mobility Cross Cluster, on page 413
• Extension Mobility Roaming Across Clusters, on page 451
• Hold Reversion, on page 465
• Accessing Hunt Groups, on page 471
• Malicious Call Identification, on page 479
• Call Transfer, on page 489
• External Call Transfer Restrictions, on page 503
Prime Line Support

- Prime Line Support Overview, on page 313
- Prime Line Support Prerequisites, on page 313
- Prime Line Support Configuration Task Flow, on page 313
- Prime Line Support Interactions, on page 315
- Prime Line Support Troubleshooting, on page 316

Prime Line Support Overview

You can configure the Prime Line Support in Cisco Unified CM Administration so that when the phone is off-hook and receives a call on any line, the system always chooses the primary line for the call.

Prime Line Support Prerequisites

The following devices are compatible with the Prime Line Support feature:
Cisco Unified IP Phone 7900 Series, 8900 Series, and 9900 Series
For more information on the supported devices, see the latest version of Cisco Unified IP Phone Guide and Cisco Unified IP Phone Administration Guide.

Prime Line Support Configuration Task Flow

To configure the Prime Line Support feature for either the Cisco CallManager service or devices and device profiles, perform one of the following procedures.

Before you begin

- Review Prime Line Support Prerequisites, on page 313.
### Configure Clusterwide Prime Line Support

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Configure Clusterwide Prime Line Support, on page 314</th>
<th>Purpose</th>
<th>(Optional) Configure the Prime Line Support feature for the Cisco CallManager service, which applies to the entire cluster.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure Prime Line Support for Devices, on page 315</td>
<td>Purpose</td>
<td>(Optional) Configure the Prime Line Support feature for specific devices within the cluster, if you do not want to enable the feature clusterwide.</td>
</tr>
<tr>
<td>Note</td>
<td>When you configure this parameter, going off-hook makes only the first line active on the phone, even when a call rings on another line on the phone. So the call does not get answered on the other line.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Configure Clusterwide Prime Line Support

**Procedure**

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

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

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

**Step 4**
From the **Always Use Prime Line** clusterwide service parameter, choose one of the following options from the drop-down list:

- **True**: When a phone goes off-hook, the primary line gets chosen and becomes the active line.
- **False**: When a phone goes off-hook, the IP phone automatically chooses an available line as the active line.

The default value for this service parameter is **False**.

**Step 5**
For this change to take effect on the SIP phones, click the **ApplyConfig** button in Cisco Unified CM Administration (for example, on the **Device Configuration** window, the **Device Pool Configuration** window, or any other window on which ApplyConfig is an option).

**Note**
If the new configuration is not applied on the SIP phones, the SIP Prime Line Support feature changes will not be implemented until the next reset of the Cisco CallManager service or reset of each affected device.

**What to do next**

(Optional) **Configure Prime Line Support for Devices, on page 315**
Configure Prime Line Support for Devices

Before you begin
(Optional) Configure Clusterwide Prime Line Support, on page 314

Procedure

**Step 1** Log in to Cisco Unified CM Administration, and choose **Device > Common Phone Profile**.

**Step 2** From the **Find and List** window, choose the phone for which you want to change the Always Use Prime Line setting.

The **Phone Configuration** window appears.

**Step 3** From the **Always Use Prime Line** 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 is 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.

**Step 4** Click **Save**.

Prime Line Support Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Always Use Prime Line</td>
<td>If you select <strong>On</strong> for the <strong>Always Use Prime Line</strong> parameter in the <strong>Device Profile</strong> or <strong>Default Device Profile Configuration</strong> window, a Cisco Extension Mobility user can use this feature after logging in to the device that supports Cisco Extension Mobility.</td>
</tr>
<tr>
<td>Maximum Number of Calls and Busy Trigger Settings</td>
<td>When the phone already has a call on a line, Cisco Unified Communications Manager uses the configuration for the Maximum Number of Calls and Busy Trigger settings to determine how to route the call.</td>
</tr>
<tr>
<td>Auto Answer</td>
<td>If you choose the Auto Answer with Headset option or Auto Answer with Speakerphone option from the <strong>Auto Answer</strong> drop-down list in Cisco Unified CM Administration, the Auto Answer configuration overrides the configuration for the <strong>Always Use Prime Line</strong> parameter.</td>
</tr>
</tbody>
</table>
Prime Line Support Troubleshooting

Prime Line Support Does Not Work When Set To True

**Problem** When the cluster-wide service parameter **Always use Prime Line** is set to **True** and the IP phone goes off-hook, the primary line becomes the active line. Even if a call rings on the second line, when the user goes off-hook, it activates only the first line. The phone does not answer the call on the second line. However, when IP phones with multiple line appearances are used with the 7.1.2 phone load, the phone does not use the primary line when a second line rings. If the user picks up the handset, the phone answers the call on the second line.

**Solution** Press the line button for the primary line so that the secondary line is not engaged when a call is initiated.

Unable To Answer Inbound Calls

**Problem** The users are unable to automatically answer inbound calls after they go off-hook on IP phones, and must press the Answer softkey to answer the calls.

**Solution** To resolve the problem, perform the following procedure:

1. In Cisco Unified CM Administration, choose **System > Service Parameters**.
2. From the Server drop-down list, choose the server that is running the Cisco CallManager service.
3. From the Service drop-down list, choose **Cisco CallManager**.
4. In Cluster wide parameters (Device - phone), set **Always Use Prime Line** to **False**.

Inbound Calls Are Answered Automatically

**Problem** When an inbound call is received on a shared line of an IP phone, the call is answered immediately as the handset is lifted, without the option to either answer the call or make an outbound call. This behavior does not change even though **Auto Line Select** is set to disabled.

**Solution** To resolve the problem, perform the following procedure:

1. In Cisco Unified CM Administration, choose **System > Service Parameters**.
2. From the Server drop-down list, choose the server that is running the Cisco CallManager service.
3. From the Service drop-down list, choose **Cisco CallManager**.
4. In Cluster wide parameters (Device - phone), set **Always Use Prime Line** to **False**.
Call Forwarding

Call Forwarding Overview

As a user, you can configure a Cisco Unified IP Phone to forward calls to another phone. The following call forwarding types are supported:

- **Call Forward No Bandwidth**—Forwards calls when a call to a directory number fails due to insufficient bandwidth, and provides forwarding functionality to an Automated Alternate Routing (AAR) destination using public switched telephone network (PSTN) as the alternate route or to a voicemail system.

- **Call Forward with Alternate Destination**—Forwards calls when a call to a directory number and the forwarded destination are not answered. The call gets diverted to an alternate destination as a last resort. This Call Forwarding type is also referred to as “MLPP Alternate Party destination.”

- **Call Forward All (CFA)**—Forwards all calls to a directory number.

- **Call Forward Busy (CFB)**—Forwards calls only when the line is in use and the configured Call Forward Busy trigger value is reached.

- **Call Forward No Answer (CFNA)**—Forwards calls when the phone is not answered after the configured No Answer Ring Duration timer is exceeded or the destination is unregistered.

- **Call Forward No Coverage (CFNC)**—Forwards calls when the hunt list is exhausted or timed out, and the associated hunt-pilot for coverage specifies “Use Personal Preferences” for its final forwarding.

- **Call Forward Unregistered (CFU)**—Forwards calls when the phone is unregistered due to a remote WAN link failure, and provides automated rerouting through the Public Switched Telephone Network (PSTN). Calls can also be forwarded based on the type of caller: internal or external.

- **CFA Destination Override**—Forwards calls when the user to whom calls are being forwarded (the target) calls the user whose calls are being forwarded (the initiator). The phone of the initiator rings instead of call forwarding back to the target.
Call Forward All, Including CFA Loop Prevention and CFA Loop Breakout

Call Forward All (CFA) allows a phone user to forward all calls to a directory number.

You can configure CFA for internal and external calls and can forward calls to a voicemail system or a dialed destination number by configuring the calling search space (CSS). Cisco Unified Communications Manager includes a secondary Calling Search Space configuration field for CFA. The secondary CSS for CFA combines with the existing CSS for CFA to allow support of the alternate CSS system configuration. When you activate CFA, only the primary and secondary CSS for CFA are used to validate the CFA destination and redirect the call to the CFA destination. If these fields are empty, the null CSS is used. Only the CSS fields that are configured in the primary CSS for CFA and secondary CSS for CFA fields are used. If CFA is activated from the phone, the CFA destination is validated by using the CSS for CFA and the secondary CSS for CFA, and the CFA destination gets written to the database. When a CFA is activated, the CFA destination always gets validated against the CSS for CFA and the secondary CSS for CFA.

Cisco Unified Communications Manager prevents CFA activation on the phone when a CFA loop is identified. For example, Cisco Unified Communications Manager identifies a call forward loop when the user presses the CFwdALL softkey on the phone with directory number 1000 and enters 1001 as the CFA destination, and 1001 has forwarded all calls to directory number 1002, which has forwarded all calls to directory number 1003, which has forwarded all calls to 1000. In this case, Cisco Unified Communications Manager identifies that a loop has occurred and prevents CFA activation on the phone with directory number 1000.

Tip
If the same directory number exists in different partitions, for example, directory number 1000 exists in partitions 1 and 2, Cisco Unified Communications Manager allows the CFA activation on the phone.

CFA loops do not affect call processing because Cisco Unified Communications Manager supports CFA loop breakout, which ensures that if a CFA loop is identified, the call goes through the entire forwarding chain, breaks out of the Call Forward All loop, and the loop is completed as expected, even if CFNA, CFB, or other forwarding options are configured along with CFA for one of the directory numbers in the forwarding chain.

For example, the user for the phone with directory number 1000 forwards all calls to directory number 1001, which has forwarded all calls to directory number 1002, which has forwarded all calls to directory number 1000, which creates a CFA loop. In addition, directory number 1002 has configured CFNA to directory number 1004. The user at the phone with directory number 1003 calls directory number 1000, which forwards to 1001, which forwards to 1002. Cisco Unified Communications Manager identifies a CFA loop, and the call, which breaks out of the loop, tries to connect to directory number 1002. If the No Answer Ring Duration timer expires before the user for the phone with directory number 1002 answers the call, Cisco Unified Communications Manager forwards the call to directory number 1004.

For a single call, Cisco Unified Communications Manager may identify multiple CFA loops and attempt to connect the call after each loop is identified.
# Call Forwarding Configuration Task Flow

## 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>Configure Partitions for Call Forwarding, on page 319</td>
<td>Administrators can configure partitions to restrict Call Forwarding to certain numbers based on the design criteria and requirements.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Calling Search Space for Call Forwarding, on page 321</td>
<td>Administrators can configure calling search spaces to restrict Call Forwarding to certain numbers based on the design criteria and requirements.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires, on page 322</td>
<td>You can forward a call when hunting fails (that is, when hunting is terminated without any hunt party answering, either because no hunt number from the list picked up or because the hunt timer timed out).</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure Call Forward No Bandwidth, on page 324</td>
<td>You can forward a call to an Automated Alternate Routing (AAR) destination using public switched telephone network (PSTN) as the alternate route or to a voicemail system when a call to a called directory number fails due to insufficient bandwidth.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure Call Forward Alternate Destination, on page 325</td>
<td>You can forward calls that go unanswered to the directory number and the forwarded destination. Calls will get diverted to an alternate destination as a last resort.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure Other Call Forwarding Types, on page 326</td>
<td>You can configure additional forwarding types such as CFA, CFB, CFNA, CFNC, and CFU. You can configure all these forwarding types from the Directory Number Configuration window.</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Enable Destination Override for Call Forwarding, on page 336</td>
<td>Administrators can override the CFA when the target of the CFA calls the initiator of the CFA. This allows the CFA target can reach the initiator for important calls.</td>
</tr>
</tbody>
</table>

## Configure Partitions for Call Forwarding

Configure partitions to create a logical grouping of directory numbers (DNs) and route 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. You can configure multiple partitions.
Configure partitions to restrict call forwarding to certain numbers based on your design criteria and requirements.

**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.

**What to do next**

Configure Calling Search Space for Call Forwarding, on page 321

**Related Topics**

Partition Name Guidelines for Call Forwarding, on page 320

**Partition Name Guidelines for Call Forwarding**

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.
Configure Calling Search Space for Call Forwarding

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. Configure calling search spaces to restrict Call Forwarding to certain numbers based on your design criteria and requirements.

Before you begin
Configure Partitions for Call Forwarding, on page 319

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.

Table 24: 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 Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires

The concept of hunting differs from that of call forwarding. Hunting allows Cisco Unified Communications Manager to extend a call to one or more lists of numbers, where each list specifies a hunting order that is chosen from a fixed set of algorithms. 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. (The next hunt party varies depending on the current hunt algorithm.) Hunting then ignores the Call Forward No Answer (CFNA), Call Forward Busy (CFB), or Call Forward All (CFA) configured values for the attempted party.

Call Forwarding allows detailed control as to how to extend (divert or redirect) a call when a called party fails to answer, or is busy and hunting is not taking place. For example, if the CFNA value for a line is set to a hunt-pilot number, a call to that line that is not answered diverts to the hunt-pilot number and begins hunting.

Before you begin

Configure Calling Search Space for Call Forwarding, on page 321

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Hunt Pilot. The Find and List Hunt Pilots window is displayed.

Step 2 Click Find. A list of configured Hunt Pilots is displayed.

Step 3 Choose the pattern for which you want to configure call treatment when hunting fails. The Hunt Pilot Configuration window is displayed.

Step 4 Configure the fields in the Hunt Pilot Configuration for the Hunt Call Treatment Settings area. See the Related Topics section for more information about the fields and their configuration options.

Step 5 Click Save.

What to do next

Configure Call Forward No Bandwidth, on page 324

Related Topics

Hunt Call Treatment Fields for Call Forwarding, on page 323
Hunt Call Treatment Fields for Call Forwarding

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hunt Call Treatment Settings</td>
<td></td>
</tr>
<tr>
<td>Note</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Hunt No Answer</td>
<td>When the call that is distributed through the hunt list is not answered in a specific period of time, this field specifies the destination to which the call gets forwarded. Choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• Do Not Forward Unanswered Calls</td>
</tr>
<tr>
<td></td>
<td>• Forward Unanswered Calls to</td>
</tr>
<tr>
<td></td>
<td>• Destination — Enter a directory number to which calls must be forwarded to.</td>
</tr>
<tr>
<td></td>
<td>• Calling Search Space — Choose a calling search space from the drop-down list which applies to all devices that use this directory number.</td>
</tr>
<tr>
<td></td>
<td>• Maximum Hunt Timer — Enter a value (in seconds) that specifies the maximum time for hunting without queuing. Valid values are 1 to 3600. The default value is 1800 seconds (30 minutes).</td>
</tr>
<tr>
<td></td>
<td>This timer is canceled if either a hunt member answers the call or the hunt list gets exhausted before the timer expires. If you do not specify a value for this timer, hunting continues until a hunt member answers or the hunt list is exhausted. If neither event takes place, hunting continues for 30 minutes, after which the call is received for final treatment.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Note</td>
<td>If hunting exceeds the number of hops that the Forward Maximum Hop Count service parameter specifies, hunting expires before the 30 minute maximum hunt timer value, and the caller receives a reorder tone.</td>
</tr>
</tbody>
</table>
### Configure Call Forward No Bandwidth

**Before you begin**

Configure Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires, on page 322

**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>Call Routing &gt; Directory Number Configuration.</strong> The <strong>Find and List Directory Numbers</strong> window is displayed.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click <strong>Find.</strong> A list of configured directory numbers is displayed.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Choose the directory number for which you want to configure call forward when there is insufficient bandwidth. The <strong>Directory Number Configuration</strong> window is displayed.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure the fields In the <strong>AAR Settings</strong> area. See the Related Topics section for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click <strong>Save.</strong></td>
</tr>
</tbody>
</table>

**What to do next**

Configure Call Forward Alternate Destination, on page 325

**Related Topics**

- Directory Number Configuration Fields for Call Forwarding, on page 325
Directory Number Configuration Fields for Call Forwarding

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Voice Mail</strong></td>
<td>Check this check box to forward the call to the voicemail.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When you check this box, Cisco Unified Communications Manager ignores the values in the <strong>Destination</strong> and <strong>Calling Search Space</strong> fields.</td>
</tr>
<tr>
<td><strong>AAR Destination Mask</strong></td>
<td>Enter a destination mask to determine the AAR destination to dial instead of using the external phone number mask.</td>
</tr>
<tr>
<td><strong>AAR Group</strong></td>
<td>Choose an AAR group from the drop-down list. It provides the prefix digits that are used to route calls that are otherwise blocked due to insufficient bandwidth. If you choose None, the server does not attempt to reroute the blocked calls. You can also configure this value in the <strong>Precedence Alternate Party Timeout</strong> service parameter from System &gt; Service Parameters.</td>
</tr>
<tr>
<td><strong>Retain this destination in the call forwarding history</strong></td>
<td>By default, the directory number configuration retains the AAR leg of the call in the call history, which ensures that the AAR forward to voicemail system will prompt the user to leave a voice message. If you check the check box, the AAR leg of the call will be present in the call forwarding history.</td>
</tr>
</tbody>
</table>

Configure Call Forward Alternate Destination

Before you begin

Configure Call Forward No Bandwidth, on page 324

Procedure

**Step 1** From Cisco Unified CM Administration, choose Call Routing > Directory Number Configuration. The Find and List Directory Numbers window is displayed.

**Step 2** Click Find. A list of configured directory numbers is displayed.

**Step 3** Choose the directory number for which you want to configure an alternate destination. The Directory Number Configuration window is displayed.

**Step 4** Configure the fields in the MLPP Alternate Party And Confidential Access Level Settings area. See the Related Topics section for more information about the fields and their configuration options.
Step 5  Click Save.

What to do next
Configure Other Call Forwarding Types, on page 326

Related Topics
MLPP Alternate Party And Confidential Access Level Settings Fields for Call Forwarding, on page 326

MLPP Alternate Party And Confidential Access Level Settings Fields for Call Forwarding

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target (Destination)</td>
<td>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 (*).</td>
</tr>
<tr>
<td>MLPP Calling Search Space</td>
<td>From the drop-down list, choose a calling search space to associate with the MLPP alternate party target (destination) number.</td>
</tr>
<tr>
<td>MLPP No Answer Ring Duration (seconds)</td>
<td>Enter the number of seconds (between 4 and 60) after which an MLPP precedence call will be directed to this directory number alternate party, if this directory number and its Call Forward destination have not answered the precedence call. You can also configure this value in the Precedence Alternate Party Timeout service parameter from System &gt; Service Parameters from Cisco Unified CM Administration.</td>
</tr>
</tbody>
</table>

Configure Other Call Forwarding Types

You can configure Call Forward All (CFA), Call Forward Busy (CFB), Call Forward No Answer (CFNA), Call Forward No Coverage (CFNC), and Call Forward Unregistered (CFU) from the Directory Number Configuration window.

Before you begin

- For Call Forwarding functionality to work as intended, Cisco recommends that for the configured phones and the directory numbers in various partitions, the Call Forward Calling Search Spaces also be configured or else the forwarding may fail. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call.

- Configure Call Forward Alternate Destination, on page 325
Procedure

**Step 1** From Cisco Unified Communications Manager, choose **Call Routing > Directory Number Configuration**. The **Find and List Directory Numbers** window is displayed.

**Step 2** Configure the **Call Forwarding and Call Pickup Settings** fields in the **Directory Number Configuration** window to configure CFA, CFB, CFNA, CFNC, and CFU. See the Related Topics section for information about the fields and their configuration options.

**Step 3** Click **Save**.

What to do next

Enable Destination Override for Call Forwarding, on page 336

Related Topics

Call Forwarding Fields, on page 327

### Call Forwarding Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forward and Call Pickup Settings</td>
<td></td>
</tr>
</tbody>
</table>
### Calling Search Space Activation Policy

Three possible values exist for this option:

- **Use System Default**—The CFA CSS Activation Policy service parameter determines which Forward All Calling Search Space to use for Call Forwarding. If the CFA CSS Activation Policy service parameter is set to **With Configured CSS**, then Forward All Calling Search Space and secondary Calling Search Space for Forward All will be used for Call Forwarding. This is the default setting.

- **With Configured CSS**—The Forward All Calling Search Space that is explicitly configured in the Directory Number Configuration window controls the Forward All activation and Call Forwarding. If the Forward All Calling Search Space is set to **None**, no CSS is configured for Forward All. A Forward All activation attempt to any directory number with a partition will fail. No change in the Forward All Calling Search Space and secondary Calling Search Space for Forward All occurs during the Forward All activation.

- **With Activating Device/Line CSS**—A combination of the Directory Number Calling Search Space and Device Calling Search Space controls the Forward All activation and Call Forwarding without explicitly configuring a Forward All Calling Search Space.

  When Forward All is activated from the phone, the Forward All Calling Search Space and secondary Calling Search Space for Forward All automatically gets populated with the Directory Number Calling Search Space and Device Calling Search Space for the activating device.

  If the Forward All Calling Search Space is set to **None**, and when Forward All is activated through the phone, the combination of Directory Number Calling Search Space and activating Device Calling Search Space controls the Forward All attempt.

### CFA CSS Activation Policy

Ensure that you configure this service parameter correctly for Forward All to work as intended in the Service Parameter Configuration window. The service parameter includes two possible values:

- **With Configured CSS**—The primary and secondary CFA Calling Search Space controls the Call Forwarding attempt.

- **With Activating Device/Line CSS**—The primary and secondary CFA Calling Search Space is updated with primary Line Calling Search Space and activating Device Calling Search Space.

### Roaming

When a device is roaming in the same device mobility group, Cisco Unified Communications Manager uses the Device Mobility CSS to reach the local gateway. If a user sets Call Forward All at the phone, the CFA CSS is set to **None**, and the CFA CSS Activation Policy is set to **With Activating Device/Line CSS**, then:

- The Device CSS and Line CSS is used as the CFA CSS when the device is in its home location.

- If the device is roaming within the same device mobility group, the Device Mobility CSS from the Roaming Device Pool and the Line CSS is used as the CFA CSS.

- If the device is roaming within a different device mobility group, the Device CSS and Line CSS is used as the CFA CSS.
### Call Forwarding Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All</td>
<td>The fields in this row of fields specify the Call Forwarding treatment for calls to this directory number if the directory number is set to forward all calls. The value in the <strong>Calling Search Space</strong> field is used to validate the Forward All destination that is entered when the user activates Call Forward All from the phone. This field is also used to redirect the call to the Call Forward All destination. Configure the following values:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Voice Mail</strong>—Check this check box to use the value that is set in the <strong>Voice Mail Profile Configuration</strong> window.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When this check box is checked, Cisco Unified Communications Manager ignores the values in the <strong>Destination</strong> and <strong>Calling Search Space</strong> fields.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Destination</strong>—This field indicates the directory number to which all calls are forwarded. Use any dialable phone number, including an outside destination.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Calling Search Space</strong>—This value applies to all devices that use this directory number.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Forward Maximum Hop Count</strong>—Configure this parameter from the Cisco Unified CM Administrator, choose <strong>System &gt; Service Parameters</strong>. This service parameter specifies the maximum number of times that a single call can be forwarded or diverted, and has special considerations for QSIG calls. For an incoming QSIG call, the maximum value is 15 (per ISO specifications); if you specify a greater value in this field, the specified value will apply to non-QSIG calls and for an incoming QSIG call, the call will only divert a maximum of 15 times. When QSIG trunks are configured, Cisco recommends setting this parameter to 15. For example, if the value of this parameter is seven, and a Call Forward All chain occurs consecutively from directory numbers 1000 to 007, which comprises seven hops, Cisco Unified Communications Manager prevents a phone user with directory number 2000 from activating CFA to directory number 1000, because no more than seven forwarding hops are supported for a single call.</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td>Because Call Forwarding is a line-based feature, in cases where the Device Calling Search Space is unknown, the system uses only the Line Calling Search Space to forward the call. If the Line Calling Search Space is restrictive and not routable, the forward attempt fails. Addition of a secondary calling search space for Call Forward All provides a solution to enable forwarding. The primary calling search space for Call Forward All and secondary calling search space for Call Forward All get concatenated (primary CFA CSS + secondary CFA CSS). Cisco Unified Communications Manager uses this combination to validate the CFA destination and to forward the call.</td>
</tr>
</tbody>
</table>
### Call Forwarding Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Busy Internal</td>
<td>The fields in this row of fields specify the forwarding treatment for internal calls to this directory number if the directory number is busy. The values in the <strong>Destination</strong> and the <strong>Calling Search Space</strong> fields are used to redirect the call to the forward destination. Configure the following values:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Voice Mail</strong>—Check this checkbox to use the configured values in the <strong>Voice Mail Profile Configuration</strong> window for internal calls.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When this check box is checked, the calling search space of the voicemail pilot is used. Cisco Unified Communications Manager ignores the values in the <strong>Destination</strong> and the <strong>Calling Search Space</strong> fields.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When this check box is checked for internal calls, the system automatically checks the <strong>Voice Mail</strong> check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the <strong>Voice Mail</strong> check box for external calls.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Destination</strong>—This field indicates the Call Forward Busy destination for internal calls. Use any dialable phone number, including an outside destination.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When you enter a destination value for internal calls, the system automatically copies this value to the <strong>Destination</strong> field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the <strong>Destination</strong> field for external calls.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Calling Search Space</strong>—The Forward Busy Internal Calling Search Space is used to forward the call to the Forward Busy Internal destination. It applies to all devices that use this directory number.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the <strong>Calling Search Space</strong> field is set to <strong>None</strong>, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward Busy Calling Search Space. If you do not configure the Forward Busy Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the <strong>Calling Search Space</strong> field for external calls.</td>
</tr>
</tbody>
</table>

The Call Forward Busy trigger is configured for each line appearance and cannot exceed the maximum number of calls that are configured for a line appearance. The Call Forward Busy trigger determines how many active calls exist on a line before the Call Forward Busy setting is activated (for example, ten calls).

**Tip** Keep the busy trigger slightly lower than the maximum number of calls so that users can make outgoing calls and perform transfers.

**Tip** If a call gets forwarded to a directory number that is busy, the call is not completed.
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Forward Busy External | The fields in this row of fields specify the forwarding treatment for external calls to this directory number if the directory number is busy. The **Destination** and **Calling Search Space** fields is used to redirect the call to the forward destination. Configure the following values:  
  • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window for external calls.  
    **Note** When this check box is checked, the calling search space of the voicemail pilot is used. Cisco Unified Communications Manager ignores the values in the **Destination** and the **Calling Search Space** fields.  
    **Note** When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the **Voice Mail** check box for external calls.  
  • **Destination**—This field indicates the Call Forward Busy destination for external calls. Use any dialable phone number, including an outside destination.  
    **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls.  
  • **Calling Search Space**—The Forward Busy External Calling Search Space forwards the call to the Forward Busy External destination. It applies to all devices that use this directory number.  
    **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward Busy Calling Search Space. If you do not configure the Forward Busy Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails.  
    **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls. |
The fields in this row of fields specify the forwarding treatment for internal calls to this directory number if the directory number does not answer. The **Destination** and **Calling Search Space** fields are used to redirect the call to the forward destination.

Configure the following values:

- **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window.

  **Note** When this check box is checked, the calling search space of the voicemail pilot is used. **Cisco Unified Communications Manager** ignores the values in the **Destination** and the **Calling Search Space** fields.

- **Destination**—This field indicates the directory number to which an internal call is forwarded when the call is not answered. Use any dialable phone number, including an outside destination.

  **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls.

- **Calling Search Space**—The Forward No Answer Internal Calling Search Space is used to forward the call to the Forward No Answer Internal destination. It applies to all devices that use this directory number.

  **Note** If the system is using partitions and calling search spaces, **Cisco recommends** that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward No Answer destination, you should also configure the Forward No Answer Calling Search Space. If you do not configure the Forward No Answer Calling Search Space and the Forward No Answer destination is in a partition, the forward operation fails.

  **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward No Answer Internal</td>
<td>The fields in this row of fields specify the forwarding treatment for internal calls to this directory number if the directory number does not answer. The <strong>Destination</strong> and <strong>Calling Search Space</strong> fields are used to redirect the call to the forward destination. Configure the following values:</td>
</tr>
</tbody>
</table>

- **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window.

  **Note** When this check box is checked, the calling search space of the voicemail pilot is used. **Cisco Unified Communications Manager** ignores the values in the **Destination** and the **Calling Search Space** fields.

- **Destination**—This field indicates the directory number to which an internal call is forwarded when the call is not answered. Use any dialable phone number, including an outside destination.

  **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls.

- **Calling Search Space**—The Forward No Answer Internal Calling Search Space is used to forward the call to the Forward No Answer Internal destination. It applies to all devices that use this directory number.

  **Note** If the system is using partitions and calling search spaces, **Cisco recommends** that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward No Answer destination, you should also configure the Forward No Answer Calling Search Space. If you do not configure the Forward No Answer Calling Search Space and the Forward No Answer destination is in a partition, the forward operation fails.

  **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls.
The fields in this row of fields specify the forwarding treatment for external calls to this directory number if the directory number does not answer. The Destination and Calling Search Space fields are used to redirect the call to the forward destination.

Configure the following values:

- **Voice Mail**—Check this check box to use the configured values in the Voice Mail Profile Configuration window.
  
  **Note** When this check box is checked, the calling search space of the voicemail pilot is used. Cisco Unified Communications Manager ignores the values in the Destination and the Calling Search Space fields.

  **Note** When this check box is checked for internal calls, the system automatically checks the Voice Mail check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the Voice Mail check box for external calls.

- **Destination**—This field indicates the directory number to which an external call is forwarded when the call is not answered. Use any dialable phone number, including an outside destination.
  
  **Note** When you enter a destination value for internal calls, the system automatically copies this value to the Destination field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the Destination field for external calls.

- **Calling Search Space**—The Forward No Answer External Calling Search Space is used to forward the call to the Forward No Answer External destination. It applies to all devices that use this directory number.
  
  **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the Calling Search Space field is set to None, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward No Answer Calling Search Space. If you do not configure the Forward No Answer Calling Search Space and the Forward No Answer destination is in a partition, the forward operation fails.

  **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the Calling Search Space field for external calls.
The Destination and Calling Search Space fields are used to redirect the call to the forward destination.

Configure the following values:

- **Voice Mail**—Check this box to use the configured values in the Voice Mail Profile Configuration window.

  **Note** When this box is checked, Cisco Unified Communications Manager ignores the values in the Destination and Calling Search Space fields. When this box is checked for internal calls, the system automatically checks the Voice Mail check box for external calls. If you do not want external calls to forward to the voicemail system, you must uncheck the Voice Mail check box for external calls.

- **Destination**—This field specifies the directory number to which an internal nonconnected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination.

  **Note** When you enter a destination value for internal calls, the system automatically copies this value to the Destination field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the Destination field for external calls.

- **Calling Search Space**—The Forward No Coverage Internal Calling Search Space is used to forward the call to the Forward No Coverage Internal destination. This value applies to all devices that use this directory number.

  **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the Calling Search Space field is set to None, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward No Coverage Calling Search Space. If you do not configure the Forward No Coverage Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails.

  **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the Calling Search Space field for external calls.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Forward No Coverage Internal | The Destination and Calling Search Space fields are used to redirect the call to the forward destination. Configure the following values:  
- **Voice Mail**—Check this box to use the configured values in the Voice Mail Profile Configuration window.  
  **Note** When this box is checked, Cisco Unified Communications Manager ignores the values in the Destination and Calling Search Space fields. When this box is checked for internal calls, the system automatically checks the Voice Mail check box for external calls. If you do not want external calls to forward to the voicemail system, you must uncheck the Voice Mail check box for external calls.  
- **Destination**—This field specifies the directory number to which an internal nonconnected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination.  
  **Note** When you enter a destination value for internal calls, the system automatically copies this value to the Destination field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the Destination field for external calls.  
- **Calling Search Space**—The Forward No Coverage Internal Calling Search Space is used to forward the call to the Forward No Coverage Internal destination. This value applies to all devices that use this directory number.  
  **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the Calling Search Space field is set to None, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward No Coverage Calling Search Space. If you do not configure the Forward No Coverage Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails.  
  **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the Calling Search Space field for external calls. |
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward No Coverage External</td>
<td>The <strong>Destination</strong> and <strong>Calling Search Space</strong> fields are used to redirect the call to the forward destination. Specify the following values:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Voice Mail</strong> — Check this check box to use the configured values in the <strong>Voice Mail Profile Configuration</strong> window.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When this check box is checked, Cisco Unified Communications Manager ignores the values in the <strong>Destination</strong> and the <strong>Calling Search Space</strong> fields. When this check box is checked for internal calls, the system automatically checks the <strong>Voice Mail</strong> check box for external calls. If you do not want external calls to forward to the voicemail system, you must uncheck the <strong>Voice Mail</strong> check box for external calls.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Destination</strong> — This field specifies the directory number to which an internal non-connected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination. <strong>Note</strong> When you enter a destination value for internal calls, the system automatically copies this value to the <strong>Destination</strong> field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the <strong>Destination</strong> field for external calls.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Calling Search Space</strong> — The Forward No Coverage External Calling Search Space is used to forward the call to the Forward No Coverage External destination. This value applies to all devices that use this directory number. <strong>Note</strong> If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured call forward calling search space is used to forward the call. If the <strong>Calling Search Space</strong> is <strong>None</strong>, the forward operation may fail if the system is using partitions and calling search spaces. For example, if you configure the Forward No Coverage destination, you should also configure the Forward No Coverage Calling Search Space. If you do not configure the Forward No Coverage Calling Search Space, and the Forward No Coverage destination is in a partition, the forward operation may fail. <strong>Note</strong> When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, choose a different value in the <strong>Calling Search Space</strong> field for external calls.</td>
</tr>
<tr>
<td>Forward on CTI Failure</td>
<td>This field applies only to CTI route points and CTI ports. The fields in this row specify the forwarding treatment for external calls to this CTI route point or CTI port if the CTI route point or CTI port fails. Configure the following values:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Voice Mail</strong> — Check this check box to use the configured values in the <strong>Voice Mail Profile Configuration</strong> window.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> When this check box is checked, Cisco Unified Communications Manager ignores the values in the <strong>Destination</strong> and <strong>Calling Search Space</strong> fields.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Destination</strong> — This field specifies the directory number to which an internal non-connected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Calling Search Space</strong> — This value applies to all devices that use this directory number.</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>This field applies to unregistered internal DN calls. The calls are rerouted to a specified destination or voicemail. You must also specify the maximum number of forwards in the <strong>Service Parameters Configuration</strong> window for a directory number in the <strong>Max Forward UnRegistered Hops to DN</strong> service parameter. <strong>Note</strong> This parameter specifies the maximum number of forward unregistered hops that are allowed for a directory number at the same time. This parameter limits the number of times the call can be forwarded due to unregistered DN when a forwarding loop occurs. Use this count to stop forward loops for external calls that have been Call Forward Unregistered. Cisco Unified Communications Manager terminates the call when the value that is specified in this service parameter is exceeded.</td>
</tr>
</tbody>
</table>
Enable Destination Override for Call Forwarding

When you enable the destination override for call forwarding, Cisco Unified Communications Manager ignores the CFA destination when it matches the calling party number. The override applies to both internal and external calls.

In cases where the calling party number has been transformed, the calling party number does not match the CFA destination, no override occurs.

Before you begin

Configure Other Call Forwarding Types, on page 326

Procedure

- **Step 1** From Cisco Unified CM Administration, choose System > Service Parameters. The Service Parameter Configuration window is displayed.
- **Step 2** In the Clusterwide Parameters (Feature - Hold Reversion) area, set the CFA Destination Override service parameter value to True.

### Table: Receiving Calls

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Unregistered External</td>
<td>This field applies to unregistered external DN calls. The calls are rerouted to a specified destination or voicemail.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> You must also specify the maximum number of forwards in the Service Parameters Configuration window for a directory number in the Max Forward UnRegistered Hops to DN service parameter. This parameter specifies the maximum number of forward unregistered hops that are allowed for a directory number at the same time. This parameter limits the number of times the call can be forwarded due to unregistered DN when a forwarding loop occurs. Use this count to stop forward loops for external calls that have been Call Forward Unregistered. Cisco Unified Communications Manager terminates the call when the value that is specified in this service parameter is exceeded.</td>
</tr>
<tr>
<td>No Answer Ring Duration (seconds)</td>
<td>This field specifies the seconds to wait before forwarding the unanswered call to the Call Forward No Answer destination, if specified. Make sure the value that is specified in this parameter is less than the value that is specified in the T301 Timer service parameter. If the value in the Forward No Answer Timer service parameter is greater than the value that is specified in the T301 Timer service parameter, the call is not forwarded and the caller receives a busy signal. Leave this field empty if you want to set the value in the Cisco Unified Communications Manager Forward No Answer Timer service parameter.</td>
</tr>
</tbody>
</table>
Call Forwarding Interactions and Restrictions

Call Forwarding Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Back</td>
<td>Calls that are made from the CallBack notification screen will override all the Call Forward configured values on the target DN. The calls should be made before the CallBack recall timer expires, otherwise the calls will not override the Call Forward configured values.</td>
</tr>
<tr>
<td>Call Display Restrictions</td>
<td>The Connected Number Display restriction applies to all calls that originate in the system. When this value is set to True, this field interacts transparently with existing Cisco Unified Communications Manager applications, features, and call processing. The value applies to all calls that terminate inside or outside the system. The Connected Number Display is updated to show the modified number or redirected number when a call is routed to a Call Forward All or Call Forward Busy destination, or gets redirected through a call transfer or CTI application.</td>
</tr>
<tr>
<td>Do Not Disturb</td>
<td>On Cisco Unified IP Phones, the message that indicates that the Do Not Disturb (DND) feature is active takes priority over the message that indicates that the user has new voice messages. However, the message that indicates that the Call Forward All feature is active has a higher priority than DND.</td>
</tr>
<tr>
<td>External Call Control</td>
<td>External Call Control intercepts calls at the translation pattern level, while Call Forward intercepts calls at the directory number level. External Call Control has higher priority; for calls where call forward is invoked, Cisco Unified Communications Manager sends a routing query to the adjunct route server if the translation pattern has an External Call Control profile assigned to it. Call Forwarding 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 Forwarding are independent of each other; they work separately.</td>
</tr>
<tr>
<td>Extension Mobility Cross Cluster</td>
<td>Cisco Extension Mobility Cross Cluster supports Call Forwarding.</td>
</tr>
<tr>
<td>Extend and Connect</td>
<td>Extend and Connect supports Call Forward All.</td>
</tr>
</tbody>
</table>

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 Forwarding are independent of each other; they work separately.
### Call Forwarding Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
</table>
| Immediate Divert   | When the Forward No Answer field in the Directory Number Configuration window is not configured, Call Forward uses the clusterwide CFNA timer service parameter, **Forward No Answer Timer**.   
                       | If a user presses the iDivert softkey at the same time as the call is being forwarded, the call gets diverted to an assigned call forward directory number (because the amount of time set on the timer was too short), not the voicemail. To resolve this situation, set the CFNA timer service parameter to enough time (for example, 60 seconds). |
| Logical Partitioning| Cisco Unified Communications Manager performs logical partitioning policy check using the geolocation identifier information that associates with the incoming and forwarded devices. This handling applies to all types of call forwarding. |
Interaction Feature

Call Forward Busy

- You can optionally configure a preconfigured Precedence Alternate Party target for any MLPP-enabled station.

- Cisco Unified Communications Manager applies the Call Forward Busy feature to forward a precedence call in the usual manner before it applies to any Precedence Alternate Party Diversion procedures to the call.

- The system preserves precedence of calls across multiple forwarded calls.

- If the incoming precedence call is of higher precedence than the existing call, preemption occurs. Both the preempted parties in the active call receive a continuous preemption tone until the station to which the precedence call is directed hangs up. After hanging up, the station to which the precedence call is directed receives precedence ringing. The destination station connects to the preempting call when the station goes off hook.

Call Forward No Answer

- For calls of Priority precedence level and above, call processing preserves the precedence level of calls during the forwarding process and may preempt the forwarded-to user.

- If an Alternate Party is configured for the destination of a precedence call, call processing diverts the precedence call to the Alternate Party after the Precedence Call Alternate Party timeout expires. If no Alternate Party value is configured for the destination of a precedence call, call processing diverts the precedence call to the Call Forward No Answer value.

- Normally, precedence calls are routed to users and not to the voicemail system. The administrator sets the Use Standard VM Handling For Precedence Calls enterprise parameter to avoid routing precedence calls to voicemail systems.

If the incoming precedence call is of equal or lower precedence than the existing call, call processing invokes normal call-forwarding behavior. If the destination station for a precedence call is nonpreemptable (that is, not MLPP-configured), call processing invokes call-forwarding behavior.

Alternate Party Diversion (APD) comprises a special type of call forwarding. If users are configured for APD, APD takes place when a precedence call is directed to a directory number (DN) that is busy or does not answer. MLPP APD applies only to precedence calls. An MLPP APD call disables the DN Call Forward No Answer value for precedence calls.
<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Originally called party name in Placed Call History</td>
<td>When privacy is configured only in the SIP profile of the called party device and Call Forward All (CFA), or Call Forward Busy (CFB), or Call Forward Unregistered (CFUR) is enabled, the configured alerting name is displayed instead of “private”. To ensure that “private” is displayed for call forwarding, Cisco recommends that you configure the name presentation restriction in the translation pattern or the route pattern rather than in the SIP profile.</td>
</tr>
</tbody>
</table>
| Rollover Lines                  | By using call forwarding settings, you can create rollover lines for a shared line. This could be useful for some call center situations. With rollover lines, when someone dials a number (e.g. 1-800-HOTLINE), the call always is routed to a specific phone line. This may be a shared line that is shared by multiple phones. If line 1 is busy, the call rolls over to line 2, if line 2 is busy it rolls over to line 3, and so on. Line 2 or 3 become available only if line 1 is busy. This type of call functionality is possible via call forwarding busy settings and the Busy Trigger as follows:  
  • On line 1, set the Busy Trigger to 1 and configure Call Forward Busy to the second line in the chain.  
  • On line 2, set the Busy Trigger to 1 and configure Call Forward Busy to the third line in the chain  
  • Continue this for as many lines as meets your needs. |
| Secure Tone                    | Call Forward All is supported on protected phones.                                                                                                                                                          |
| Session Handoff                | When the user hands off a call, a new call gets presented on the desk phone. While the desk phone is flashing, Call Forward All is not triggered on the desk phone for the call that was handed off.                                                                 |

**Receiving Calls**

**Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1**
## Call Forwarding Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forwarding</td>
<td>• If Call Forward All activation occurs in Cisco Unified Communications Manager Administration or the Cisco Unified Communications Self Care Portal, Cisco Unified Communications Manager does not prevent the CFA loop.</td>
</tr>
<tr>
<td></td>
<td>• Cisco Unified Communications Manager prevents Call Forward All loops if CFA is activated from the phone, if the number of hops for a Call Forward All call exceeds the value that is specified for the Forward Maximum Hop Count service parameter, and if all phones in the forwarding chain have CFA activated (not CFB, CFNA, or any other call forwarding options).</td>
</tr>
<tr>
<td></td>
<td>For example, if the user with directory number 1000 forwards all calls to directory number 1001, which has CFB and CFNA configured to directory number 1002, which has CFA configured to directory number 1000, Cisco Unified Communications Manager allows the call to occur because directory number 1002 acts as the CFB and CFNA (not CFA) destination for directory number 1001.</td>
</tr>
<tr>
<td></td>
<td>• You cannot activate Call Back if you forward all calls to voicemail system.</td>
</tr>
<tr>
<td></td>
<td>• An uncommon condition in connection with the Forward No Answer Timeout exists when you press the iDivert softkey. For example, if a manager presses the iDivert softkey immediately after the Forward No Answer timeout, Call Forward forwards the call to a preconfigured directory number. However, if the manager presses the iDivert softkey before the Forward No Answer timeout, Immediate Divert diverts the call to the voicemail of the manager.</td>
</tr>
</tbody>
</table>

<p>| Immediate Divert             | When Call Forward All (CFA) and Call Forward Busy (CFB) are activated, the system does not support Immediate Divert (CFA and CFB have precedence over Immediate Divert). |
| Intercom                     | You cannot forward Intercom calls.                                                                                                           |
| Log Out of Hunt Group        | When a phone that is running SIP (7906, 7911, 7941, 7961) is logged in to hunt groups and Call Forward All is activated, the call gets presented to the phone that is running SIP. When 7940 and 7960 IP phones that are running SIP are logged in to hunt groups and Call Forward All is activated, the phone gets skipped and the next phone in the line group is rung. |</p>
<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
</table>
| Logical Partitioning | Logical partitioning handling does not take place in the following circumstances:  
  • When both the caller and forwarded devices are Voice over IP (VoIP) phones.  
  • When geolocation or a geolocation filter is not associated with any device. |
| Multilevel Precedence and Preemption (MLPP) | Multilevel Precedence and Preemption (MLPP) support for supplementary services specifies the following restrictions for Call Forwarding:  
  • 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 gets rejected, and the calling party receives 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 remains unanswered, the call is sent to the MAP target of the original 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 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 is configured for forwarding hops. If the maximum hop count is reached, the call is sent to the MAP target of the original called party, if the MAP target is 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. |
| Call Forward Classification with Call Transfer | When a call is transferred, the call classification takes on the classification of the transferred leg, rather than the original leg. For example:  
  • Incoming call from PSTN is received by a receptionist. This is an external call.  
  • The receptionist transfers the call to extension 3100. The transferred call is now an internal call.  
  • The user at extension 3100 is busy, but has Call Forward External configured to send external calls back to the receptionist. However, because the call takes on the classification of the second leg (internal), the call goes to voicemail. |
Call Pickup Overview

The Call Pickup feature allows users to answer calls that come in on a directory number other than their own.

Group Call Pickup Overview

The Group Call Pickup feature allows users to pick up incoming calls in another group. Users must dial the appropriate call pickup group number when this feature is activated from a Cisco Unified IP Phone. Use the softkey, GPickUp, for this type of call pickup. When the user invokes the Group Call Pickup phone feature while multiple calls are incoming to a pickup group, the user gets connected to the incoming call that has been ringing the longest. Depending on the phone model, the users can either use the Group Pickup programmable feature button or the Group Pickup softkey to pick up an incoming call. If Auto Group Call Pickup is not enabled, the user must press the GPickUp softkey, dial the group number of another pickup group, and answer the call to make the connection.

Other Group Pickup Overview

The Other Group Pickup feature allows users to pick up incoming calls in a group that is associated with their own group. The Cisco Unified Communications Manager automatically searches for the incoming call in the associated groups to make the call connection when the user activates this feature from a Cisco Unified IP Phone. Users use the softkey, OPickUp, for this type of call pickup. If Auto Other Group Pickup is not enabled, the user must press the softkeys, OPickUp and Answer, to make the call connection. Depending on the phone model, the users can either use the Call Pickup programmable feature button or the Call Pickup softkey to pick up an incoming call.

When more than one associated group exists, the first associated group has the highest the priority of answering calls for the associated group. For example, groups A, B, and C associate with group X, the group A has the highest priority and the group C has the lowest priority of answering calls. The group X picks up incoming call in group A, though a call may have come in earlier in group C than the incoming call in group A.
Directed Call Pickup Overview

The Directed Call Pickup feature allows a user to pick up a ringing call on a DN directly by pressing the GPickUp or Group Pickup softkeys and entering the directory number of the device that is ringing. If Auto Directed Call Pickup is not enabled, the user must press the GPickUp softkey, dial the DN of the ringing phone, and answer the call that will now ring on the user phone to make the connection. Cisco Unified Communications Manager uses the associated group mechanism to control the privilege of a user who wants to pick up an incoming call by using Directed Call Pickup. The associated group of a user specifies one or more call pickup groups that are associated to the pickup group to which the user belongs.

If a user wants to pick up a ringing call from a DN directly, the associated groups of the user must contain the pickup group to which the DN belongs. If two users belong to two different call pickup groups and the associated groups of the users do not contain the call pickup group of the other user, the users cannot invoke Directed Call Pickup to pick up calls from each other.

When the user invokes the Directed Call Pickup feature and enters a DN to pick up an incoming call, the user connects to the call that is incoming to the specified phone whether or not the call is the longest ringing call in the call pickup group to which the DN belongs. If multiple calls are ringing on a particular DN and the user invokes Directed Call Pickup to pick up a call from the DN, the user connects to the incoming call that has been ringing the specified DN the longest.

BLF Call Pickup Overview

The BLF Call Pickup feature allows Cisco Unified Communications Manager to notify a phone user when a call is waiting to be picked up from a BLF DN. The BLF call pickup initiator (the phone that picks up the call) is selected as the next available line or as a specified line. To use a specified line, the line must remain off hook before the BLF SD button is pressed. You can configure a hunt list member DN as the BLF DN to allow an incoming call to a hunt list member to be picked up by the BLF call pickup initiator. The incoming call on the hunt list member can come from the hunt list or be a directed call. The behavior in each case depends on how you configure call pickup for the hunt list member DN, the BLF DN, and the hunt pilot number. When a Call Pickup occurs with the service parameter Auto Call Pickup Enabled set to false, the phone must remain off hook or the user must press the answer key to pick up the call.

The BLF SD button on the phone can exist in any of the following states:

- **Idle**—Indicates that no call exists on the BLF DN.
- **Busy**—Indicates that at least one active call exists on the BLF DN, but no alerts exist.
- **Alert**—Indicates by flashing that at least one incoming call exists on the BLF DN.

When there is an incoming call to the BLF DN, the BLF SD button flashes on the BLF call pickup initiator phone to indicate that an incoming call to the BLF DN exists. If Auto Call Pickup is configured, the user presses the BLF SD button on the call pickup initiator phone to pick up the incoming call. If auto call pickup is not configured, the phone must remain off hook, or the user must press the answer key to pick up the call.
# Call Pickup Configuration Task Flow

## Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Configure a Call Pickup Group, on page 347</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Configure a call pickup group for each of the call pickup features that you want to use:</td>
</tr>
<tr>
<td></td>
<td>• Call Pickup</td>
</tr>
<tr>
<td></td>
<td>• Group Call Pickup</td>
</tr>
<tr>
<td></td>
<td>• Other Call Pickup</td>
</tr>
<tr>
<td></td>
<td>• Directed Call Pickup</td>
</tr>
<tr>
<td></td>
<td>• BLF Call Pickup</td>
</tr>
<tr>
<td></td>
<td>You must define groups with unique names and numbers.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Assign a Call Pickup Group to Directory Numbers, on page 347</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Assign each of the call pickup groups that you created to the directory numbers that are associated with phones on which you want to enable call pickup. Directory numbers must be assigned to a call pickup group to use this feature.</td>
</tr>
<tr>
<td></td>
<td>Repeat this procedure for each call pickup group that you create.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 3</th>
<th>Create another call pickup group and associate it with the BLF call pickup group that you created in Step 1, on page 345. You can associate a call pickup group with multiple BLF DN call pickup groups.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Perform this step if you are configuring BLF Call Pickup.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>You do not always need to create another call pickup group. For example, you can have a single call pickup group that includes both the initiator DN and the destination DN. In such cases, associate the BLF call pickup group with itself.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 4</th>
<th>Configure Partitions for Call Pickup, on page 348</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Configure partitions to create a logical grouping of directory numbers (DN) with similar reachability characteristics. You can use partitions to restrict access to call pickup groups. If you assign call pickup group numbers to a partition, only those phones that can dial numbers in that partition can use the call pickup group.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td>You must complete this procedure for directed call pickup. It is optional for other types of call pickup.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure Calling Search Space, on page 349</td>
<td>If you configure partitions, you must also configure calling search spaces. Configure calling search spaces to identify the partitions that calling devices can search when they attempt to complete a call. You must complete this procedure for directed all pickup. It is optional for other types of call pickup.</td>
</tr>
<tr>
<td>(Optional) Assign a Call Pickup Group to Hunt Pilots, on page 350</td>
<td>Assign a call pickup group to a hunt pilot DN so that users can pick up calls that are alerting in the line group members. Hunt lists that are assigned to a call pickup group can use Call Pickup, Group Call pickup, BLF Call Pickup, Other Group Pickup, and Directed call pickup.</td>
</tr>
<tr>
<td>(Optional) Configure notifications:</td>
<td>Configure notifications when other members of a pickup group receive a call. You can configure audio or visual notifications, or both.</td>
</tr>
<tr>
<td>• Configure Call Pickup Notification, on page 350</td>
<td></td>
</tr>
<tr>
<td>• Configure Call Pickup Notification for a Directory Number, on page 352</td>
<td></td>
</tr>
<tr>
<td>• Configure BLF Call Pickup Notification, on page 353</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> Configure Directed Call Pickup:</td>
<td>Before you configure directed call pickup, you must configure partitions and calling search spaces. With directed call pickup, the calling search space of the user who requests the Directed Call Pickup feature must contain the partition of the DN from which the user wants to pick up a call. Time periods and time schedules specify the times when members in the associated group are available to accept calls.</td>
</tr>
<tr>
<td>• Configure a Time Period, on page 354</td>
<td></td>
</tr>
<tr>
<td>• Configure Time Schedule, on page 355</td>
<td></td>
</tr>
<tr>
<td>• Associate a Time Schedule with a Partition, on page 355</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong> (Optional) Configure automatic call answering:</td>
<td>Enable automatic call answering and configure timers for automatic call answering.</td>
</tr>
<tr>
<td>• Configure Auto Call Pickup, on page 356</td>
<td></td>
</tr>
<tr>
<td>• Configure BLF Auto Pickup, on page 357</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong> Configure phone button templates:</td>
<td>Configure phone button templates for any of the call pickup features that you want to use:</td>
</tr>
<tr>
<td>• Configure Call Pickup Phone Button Template, on page 357</td>
<td>• Speed Dial BLF</td>
</tr>
<tr>
<td>• Associate Call Pickup Button Template with Phone, on page 358</td>
<td>• Call Pickup</td>
</tr>
</tbody>
</table>
Configure a Call Pickup Group

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose Call Routing > Call Pickup Group. The Find and List Call Pickup Groups window appears.

**Step 2**  
Click Add New. The Call Pickup Group Configuration window appears.

**Step 3**  
Configure the fields in the Call Pickup Group Configuration window. See the online help for more information about the fields and their configuration options.

Assign a Call Pickup Group to Directory Numbers

This section describes how to assign a call pickup group to a directory number. Only directory numbers that are assigned to a call pickup group can use call pickup, group call pickup, BLF call pickup, other group pickup, and directed call pickup. If partitions are used with call pickup numbers, make sure that the directory numbers that are assigned to the call pickup group have a calling search space that includes the appropriate partitions.

**Before you begin**

Configure a Call Pickup Group, on page 347

**Procedure**

**Step 1**  
Choose Device > Phone or Call Routing > Directory Number.
Configure Partitions for Call Pickup

You can restrict access to call pickup groups by assigning a partition to the call pickup group number. When this configuration is used, only the phones that have a calling search space that includes the partition with the call pickup group number can participate in that call pickup group. Make sure that the combination of partition and group number is unique throughout the system. You can create multiple partitions.

If you assign call pickup group numbers to a partition, only those phones that can dial numbers in that partition can use the call pickup group. If partitions represent tenants in a multitenant configuration, make sure that you assign the pickup groups to the appropriate partition for each tenant.

Before you begin

Assign a Call Pickup Group to Directory Numbers, on page 347

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Partition.

Step 2 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 3 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 4 To create multiple partitions, use one line for each partition entry.
Step 5 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 6 Select one of the following radio buttons to configure the **TimeZone**:

- **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 7 Click **Save**.

---

**What to do next**

Configure Calling Search Space, on page 349

---

**Configure Calling Search Space**

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.

**Before you begin**

Configure Partitions for Call Pickup, on page 348

**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.
Step 8
Click Save.

What to do next
Assign a Call Pickup Group to Hunt Pilots, on page 350

Assign a Call Pickup Group to Hunt Pilots

Only hunt lists that are assigned to a call pickup group can use Call Pickup, Group Call Pickup, BLF Call Pickup, Other Group Pickup, and Directed Call Pickup. Follow these steps to assign a call pickup group to hunt pilots:

Before you begin
Configure Calling Search Space, on page 349

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Call Routing &gt; Route/Hunt &gt; Hunt Pilot.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Enter the appropriate search criteria to find the hunt pilot that you want to assign to a call pickup group and click Find. A list of hunt pilots that match the search criteria appears.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose the hunt pilot to which you want to assign a call pickup group.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Call Pickup Group drop-down list box that appears in the Hunt Forward Settings area, choose the desired call pickup group.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

What to do next
Configure Call Pickup Notification, on page 350

Configure Call Pickup Notification

You can configure Call Pickup Notification at the system level, call pickup group level, or individual phone level.

Before you begin
Assign a Call Pickup Group to Hunt Pilots, on page 350

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configure Call Pickup Notification for a Call Pickup Group, on page 351</td>
<td>To allow the original called party to pick up the call prior to the audio and/or visual alert being sent to the pickup group.</td>
</tr>
</tbody>
</table>
Configure Call Pickup Notification for a Call Pickup Group

Before you begin
Assign a Call Pickup Group to Hunt Pilots, on page 350

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure Call Pickup Notification for a Directory Number, on page 352</td>
<td>To configure the type of audio alert to be provided when phone is idle or has an active call.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure BLF Call Pickup Notification, on page 353</td>
<td></td>
</tr>
</tbody>
</table>

Configure Call Pickup Notification for a Call Pickup Group

Before you begin
Assign a Call Pickup Group to Hunt Pilots, on page 350

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Call Pickup Group. The Call Pickup Group window appears.

Step 2 Configure the fields in the Call Pickup Group Notification Settings section in the Call Pickup Group Configuration window. See the Related Topics section for details about the fields and their configuration options.

Note Refer to Call Pickup Interactions and Restrictions, on page 363 for feature interactions and restrictions that will affect your Call Pickup configuration.

What to do next
Configure Call Pickup Notification for a Directory Number, on page 352

Related Topics
Call Pickup Notification Fields for Call Pickup, on page 351

Call Pickup Notification Fields for Call Pickup

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Pickup Group Notification Policy</td>
<td>From the drop-down list, select the notification policy. The available are options are No Alert, Audio Alert, Visual Alert, and Audio and Visual Alert.</td>
</tr>
<tr>
<td>Call Pickup Group Notification Timer</td>
<td>Enter the seconds of delay (integer in the range of 1 to 300) between the time that the call first comes into the original called party and the time that the notification to the rest of the call pickup group is sent.</td>
</tr>
</tbody>
</table>
Configure Call Pickup Notification for a Directory Number

Perform these steps to configure the type of audio notification that is provided when a phone is idle or in use.

**Before you begin**
Configure Call Pickup Notification for a Call Pickup Group, on page 351

**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 Call Routing &gt; Directory Number. The Find and List Directory Numbers window appears.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Enter the search criteria and click Find.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Click the directory number for which you want to configure the Call Pickup Notification. The Directory Number Configuration window appears.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Choose a device name in the Associated Devices pane and click the Edit Line Appearance button. The Directory Number Configuration window refreshes to show the line appearance for this DN on the device that you choose.</td>
</tr>
</tbody>
</table>
| **Step 5** | From the Call Pickup Group Audio Alert Setting(Phone Idle) drop-down list, choose one of the following:  
  - Use System Default  
  - Disable  
  - Ring Once |
| **Step 6** | From the Call Pickup Group Audio Alert Setting(Phone Active) drop-down list, choose one of the following:  
  - Use System Default  
  - Disable  
  - Beep Only |
Step 7 Click Save.

What to do next
Configure BLF Call Pickup Notification, on page 353

Configure BLF Call Pickup Notification

Before you begin
Configure Call Pickup Notification for a Directory Number, on page 352

Procedure

Step 1 In Cisco Unified CM Administration, choose System > Service Parameters.
Step 2 From the Server drop-down list box, choose the server that is running the Cisco CallManager service.
Step 3 From the Service drop-down list box, choose Cisco CallManager.
Step 4 Configure the fields from Clusterwide Parameters (Device - Phone) section in the Service Parameter Configuration window. See the Related Topics section for more information about the fields and their configuration options.

What to do next
Configure Directed Call Pickup, on page 354

Related Topics
Service Parameter Fields for BLF Call Pickup Notification, on page 353

Service Parameter Fields for BLF Call Pickup Notification

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Call Pickup Group Audio Alert Setting of Idle Station | This parameter determines the kind of audio notification that is provided when a phone is idle (not in use) and it needs to be alerted regarding an incoming call on its Call Pickup Group. Valid values are as follows:  
  - Disable  
  - Ring Once |
### Configure Directed Call Pickup

#### 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>Configure a Time Period, on page 354</td>
<td>Configure time period for members of the associated groups to your group.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Time Schedule, on page 355</td>
<td>Configure time schedule for members of the associated groups to your group.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Associate a Time Schedule with a Partition, on page 355</td>
<td>Associate time schedules with partitions to determine where calling devices search when they are attempting to complete a call during a particular time of a day.</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.
# Configure Time Schedule

## Before you begin

Configure a Time Period, on page 354

## Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Class of Control &gt; Time Period</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure the fields in the <strong>Time Period Configuration</strong> window. For more information on the fields and their configuration options, see the system Online Help.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

## What to do next

Configure Time Schedule, on page 355

## 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.

## Before you begin

Configure Time Schedule, on page 355

## Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Class of Control &gt; Partition</strong>.</th>
</tr>
</thead>
</table>
| Step 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. |
| Step 3 | Click **Save**. |
Configure Automatic Call Answering

What to do next
Configure Automatic Call Answering, on page 356

Configure Automatic Call Answering

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Auto Call Pickup, on page 356</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure BLF Auto Pickup, on page 357</td>
</tr>
</tbody>
</table>

Configure Auto Call Pickup

Auto call pickup connects the user to an incoming call. When the user presses the softkey on the phone, Cisco Unified Communications Manager locates the incoming call in the group and completes the call connection. You can automate call pickup, group pickup, other group pickup, directed call pickup, and BLF call pickup. If you do not enable automatic call answering, users must press additional softkeys or dial group numbers to complete the connection.

Before you begin
Associate a Time Schedule with a Partition, on page 355

Procedure

Step 1  In Cisco Unified CM Administration, choose, **System > Service Parameters**.
Step 2  From the Server drop-down list box, choose the server that is running the Cisco CallManager service.
Step 3  From the Service drop-down list box, choose **Cisco CallManager**.
Step 4  In the **Clusterwide Parameters (Feature – Call Pickup)** section, select **True** or **False** from the **Auto Call Pickup Enabled** drop-down list to enable or disable automatic call answering for call pickup groups.
Step 5  If the **Auto Call Pickup Enabled** service parameter is False, enter a value from 12 to 300 in the **Call Pickup No Answer Timer** field. This parameter controls the time that a call takes to get restored if the call is picked up but not answered by using call pickup, group call pickup, or other group call pickup.
Step 6  In the **Pickup Locating Timer** field, enter a value from 1 to 5. This service parameter specifies the maximum time, in seconds, for Cisco Unified Communications Manager to identify all alerting calls from all nodes in the cluster. This information is then used to help ensure that the call that has been waiting longest in the queue is delivered to the next user who presses thePickUp, GPickUp, or OPickUp softkey.
Step 7  Click **Save**.
Configure BLF Auto Pickup

Before you begin
Configure Auto Call Pickup, on page 356

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 BLF Auto Pickup</td>
<td>In Cisco Unified CM Administration, choose, <strong>System &gt; Service Parameters</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From the <strong>Server</strong> drop-down list box, choose the server that is running the Cisco CallManager service.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Service</strong> drop-down list box, choose <strong>Cisco CallManager</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure values for the following clusterwide service parameters.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• BLF Pickup Audio Alert Setting of Idle Station—Select <strong>True</strong> or <strong>False</strong> from the drop-down list to enable or disable automatic call answering for call pickup groups. The default value for this service parameter is False.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• BLF Pickup Audio Alert Setting of Busy Station—If the Auto Call Pickup Enabled service parameter is False, enter a value from 12 to 300 (inclusive). This parameter controls the time that a call takes to get restored if the call is picked up but not answered by using call pickup, group call pickup, or other group call pickup.</td>
<td></td>
</tr>
</tbody>
</table>

Configure Call Pickup Phone Buttons

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Call Pickup Phone Button Template, on page 357</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate Call Pickup Button Template with Phone, on page 358</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure BLF Speed Dial Number for the BLF Call Pickup Initiator, on page 359</td>
</tr>
</tbody>
</table>

Configure Call Pickup Phone Button Template

Follow these steps to add Call Pickup feature to the phone button template.
Before you begin
Configure Call Pickup Phone Button Template, on page 357

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.

---

**What to do next**

Configure BLF Speed Dial Number for the BLF Call Pickup Initiator, on page 359

---

**Configure BLF Speed Dial Number for the BLF Call Pickup Initiator**

**Before you begin**

Associate Call Pickup Button Template with Phone, on page 358

**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 <strong>Device &gt; Phone</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Select the phone that you want to use as the BLF call pickup initiator.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>In the <strong>Association</strong> pane, <strong>Add a new BLF SD link</strong>.</td>
<td>The Busy Lamp Field Speed Dial Configuration window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Select a <strong>Directory Number</strong> (BLF DN) that should be monitored by the BLF SD button.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Check the <strong>Call Pickup</strong> check box to use the BLF SD button for BLF Call Pickup and BLF Speed Dial. If you do not check this check box, the BLF SD button will be used only for BLF Speed Dial.</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>Click <strong>Save</strong>.</td>
<td></td>
</tr>
</tbody>
</table>

---

**What to do next**

Configure Softkeys for Call Pickup, on page 359

---

**Configure Softkeys for Call Pickup**

**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>Configure a Softkey Template for Call Pickup, on page 360</strong></td>
<td>Add the Pickup, GPickup, and OPickup softkeys to a softkey template.</td>
</tr>
<tr>
<td>Step 2</td>
<td>To <strong>Associate a Softkey Template with a Common Device Configuration, on page 361</strong>, complete the following subtasks:</td>
<td>Optional. 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.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Add a Softkey Template to Common Device Configuration, on page 362</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td>• <strong>Associate a Common Device Configuration with a Phone, on page 362</strong></td>
<td></td>
</tr>
</tbody>
</table>
### Purpose

**Command or Action**

**Purpose**

**Step 3**  
**Associate a Softkey Template with a Phone**, on page 362  
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.

---

### Configure a Softkey Template for Call Pickup

Use this procedure to make the following call pickup softkeys available:

<table>
<thead>
<tr>
<th>Softkey</th>
<th>Description</th>
<th>Call States</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Pickup (Pickup)</td>
<td>Allows you to answer a call on another extension in your group.</td>
<td>On Hook, Off Hook</td>
</tr>
<tr>
<td>Group Call Pickup (GPickup)</td>
<td>Allows you to answer a call on extension outside your group.</td>
<td>On Hook, Off Hook</td>
</tr>
<tr>
<td>Other Group Pickup (OPickup)</td>
<td>Allows you to answer a call ringing in another group that is associated with your group.</td>
<td>On Hook, Off Hook</td>
</tr>
</tbody>
</table>

### Before you begin

Configure Call Pickup Phone Buttons, on page 357

### 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.
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:

- If you modified a template that is already associated with devices, click Apply Config 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 tasks:

- Associate a Softkey Template with a Common Device Configuration, on page 361
- Associate a Softkey Template with a Phone , on page 362

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 Associate a Softkey Template with a Phone , on page 362.

Procedure

Step 1  Add a Softkey Template to Common Device Configuration, on page 362

Step 2  Associate a Common Device Configuration with a Phone, on page 362
Add a Softkey Template to 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.

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.

Associate a Softkey Template with a Phone

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.

Call Pickup Interactions and Restrictions

Call Pickup Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Plan Report</td>
<td>The route plan report displays the patterns and DN numbers that are configured in Cisco Unified Communications Manager. Use the route plan report to look for overlapping patterns and DN numbers before assigning a DN to call pickup group.</td>
</tr>
<tr>
<td>Calling search space and partitions</td>
<td>Assigning a partition to the Call Pickup Group number limits call pickup access to users on the basis of the device calling search space.</td>
</tr>
<tr>
<td>Time of Day (TOD)</td>
<td>Time of Day (TOD) parameter for members in the associated group enable them to accept calls within the same time period as their own group. TOD associates a time stamp to the calling search space and partition.</td>
</tr>
<tr>
<td>Call Accounting</td>
<td>When a call pickup occurs through auto call pickup, the system generates two call detail records (CDRs). One CDR applies to the original call that is cleared, and another CDR applies to the requesting call that is connected.</td>
</tr>
<tr>
<td></td>
<td>A CDR search returns all CDRs that match a specific time range and other search criteria. You can also search for a type of call that is associated with a particular CDR. The search result displays a call type field that indicates whether the call is a pickup call.</td>
</tr>
<tr>
<td>Call Forwarding</td>
<td>When a call pickup occurs with the service parameter Auto Call Pickup Enabled set to false, the call forward that is configured on the phone gets ignored when one of the pickup softkeys is pressed. If the call pickup requestor does not answer the call, the original call gets restored after the pickup no answer timer expires.</td>
</tr>
</tbody>
</table>

Call Pickup Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Different phone lines to different call pickup groups</td>
<td>Although you can assign different lines on a phone to different call pickup groups, Cisco does not recommend this setup because it can be confusing to users.</td>
</tr>
</tbody>
</table>
### Call Pickup Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| Call Pickup Group Number                        | • You cannot delete a call pickup group number when it is assigned to a line or DN. To determine which lines are using the call pickup group number, use Dependency Records in **Call Pickup Configuration** window. To delete a call pickup group number, reassign a new call pickup group number to each line or DN.  
  • When you update a call pickup group number, Cisco Unified Communications Manager automatically updates all directory numbers that are assigned to that call pickup group. |
| SIP Phones                                      | • The system does not support Call Pickup Notification on a few Cisco Unified IP Phones that run SIP.                                                                                                         
  • Call Pickup Notification is only supported on licensed, third-party phones that run SIP.                                                                                                                               |
| Directed Call Pickup                            | • If a device that belongs to a hunt list rings due to a call that was made by calling the hunt pilot number, users cannot use the Directed Call Pickup feature to pick up such a call.  
  • Users cannot pick up calls to a DN that belongs to a line group by using the Directed Call Pickup feature.                                                                                                           |
| BLF Pickup                                      | The system does not support Call Pickup Notification on a few Cisco Unified IP Phones that run SIP.                                                                                                                                                                   |
| Incoming Calling Party International Number Prefix - Phone | If you have configured a prefix in the “Incoming Calling Party International Number Prefix - Phone” service parameter, and an international call is placed to a member in the Call Pickup Group, the prefix does not get invoked in the calling party field if the call gets picked up by another member of the Call Pickup Group. |
Call Park Overview

The Call Park feature allows you to place a call on hold so that it can be retrieved from another phone in the Cisco Unified Communications Manager system (for example, a phone in another office or in a conference room). If you are on an active call, you can park the call to a call park extension by pressing the Park softkey. Another phone in your system can then dial the call park extension to retrieve the call.

You can define either a single directory number or a range of directory numbers for use as Call Park extension numbers. You can park only one call at each Call Park extension number.

The Call Park feature works within a Cisco Unified Communications Manager cluster, and each Cisco Unified Communications Manager node in a cluster must have Call Park extension numbers defined. You can define either a single directory number or a range of directory numbers for use as Call Park extension numbers.

Ensure that the directory number or range of numbers is unique.

Users can dial the assigned route pattern (for example, a route pattern for an intercluster trunk could be 80XX) and the Call Park number (for example, 8022) to retrieve parked calls from another Cisco Unified Communications Manager cluster. You must ensure that calling search spaces and partitions are properly configured. Call Park works across clusters.

Valid Call Park extension numbers comprise integers and the wildcard character X. You can configure a maximum of XX in a Call Park extension number (for example, 80XX), which provides up to 100 Call Park extension numbers. When a call gets parked, Cisco Unified Communications Manager chooses the next Call Park extension number that is available and displays that number on the phone.
Park Monitoring

Park Monitoring is an optional Call Park feature where Cisco Unified Communications Manager monitors the status of a parked call until a timer expires. After the timer expires, the call is forwarded to a preassigned number, sent to voicemail, or returned to the call parker. You can apply park monitoring to phone lines and to hunt pilots.

Call Park Prerequisites

If you are using call park across clusters, you must have partitions and calling search spaces configured.

Table 25: Cisco Unified IP Phones that Support Park Softkey Template and Call Park Button Template

<table>
<thead>
<tr>
<th>Phone Model</th>
<th>Supported in Softkey Template</th>
<th>Supported in Phone Button Template</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phones 6900 series (except 6901 and 6911)</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco IP Phone 7800 Series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phones 7900 series (except 7921, 7925, 7936, 7937)</td>
<td>X</td>
<td></td>
</tr>
<tr>
<td>Cisco IP Phone 8800 Series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phones 8900 series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phones 9900 series</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Cisco Unified IP Phones 7900 series (except 7906, 7911, 7921, 7925, 7936, 7937)</td>
<td></td>
<td>X</td>
</tr>
</tbody>
</table>

Note

You can configure Call Park on any line (except line 1) or button by using the programmable line key feature.

Call Park Configuration Task Flow

Before you begin

- Review Call Park Prerequisites, on page 366
## Configure Clusterwide Call Park

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Clusterwide Call Park, on page 367</td>
<td>(Optional) Configure Call Park for the entire cluster, or use the procedure in step 3 to configure Call Park on servers within the cluster</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure a Partition for Call Park, on page 368</td>
<td>Create a partition to add a Call Park Number</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure a Call Park Number, on page 369</td>
<td>Configure a Call Park Number to use Call Park across servers in a cluster.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure a Softkey Template for Call Park, on page 372</td>
<td>Add the Park softkey to a softkey template.</td>
</tr>
<tr>
<td><strong>Step 5</strong> To Associate a Softkey Template with a Common Device Configuration, on page 373, complete the following subtasks:</td>
<td>Optional. 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.</td>
</tr>
<tr>
<td>• Add a Softkey Template to a Common Device Configuration, on page 373</td>
<td></td>
</tr>
<tr>
<td>• Associate a Common Device Configuration with a Phone, on page 374</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> Associate a Softkey with a Phone, on page 374</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 7</strong> To Configure Call Park Button, on page 374, complete the following subtasks:</td>
<td></td>
</tr>
<tr>
<td>• Configure a Phone Button Template for Call Park, on page 374</td>
<td></td>
</tr>
<tr>
<td>• Associate a Button Template with a Phone, on page 375</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> Configure Park Monitoring, on page 376</td>
<td>Complete this optional task flow to add Park Monitoring to your Call Park configuration.</td>
</tr>
</tbody>
</table>

---

### Configure Clusterwide Call Park

**Procedure**

**Step 1** Choose **System > Service Parameters**.
Configure a Partition for Call Park

Configure partitions to create a logical grouping of directory numbers (DNs) and route 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. You can configure multiple partitions.

### Before you begin
(Optional) Configure Clusterwide Call Park, on page 367

### 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.

---

**Configure a Partition for Call Park**

Configure partitions to create a logical grouping of directory numbers (DNs) and route 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. You can configure multiple partitions.

### Before you begin
(Optional) Configure Clusterwide Call Park, on page 367

### Procedure

**Step 2** Select the desired node as **Server** and the service as **Cisco CallManager** (active).

**Step 3** Click the **Advanced** button.

The advanced service parameters are displayed in the window.

**Step 4** In Clusterwide Parameter (Feature-General) section set the **Enable cluster-wide Call Park Number/Ranges** to **True**.

The default value is False. This parameter determines whether the Call Park feature is implemented clusterwide or restricted to a specific Unified CM node.

**Step 5** Set the **Call Park Display Timer** for each server in a cluster that has the Cisco CallManager service and Call Park configured.

The default is 10 seconds. This parameter determines how long a Call Park number displays on the phone that parked the call.

**Step 6** Set the **Call Park Reversion Timer** for each server in a cluster that has the Cisco Unified Communications Manager service and Call Park configured.

The default is 60 seconds. This parameter determines the time that a call remains parked. When this timer expires, the parked call returns to the device that parked the call. If a hunt group member parks a call that comes through a hunt pilot, the call goes back to the hunt pilot when the Call Park Reversion Timer expires.

**Note** If you enter a Call Park Reversion Timer value that is less than the Call Park Display Timer, Call Park numbers may not display on the phone.

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

**Step 8** Restart all Cisco Unified Communications Manager services.

---

**What to do next**

Configure a Partition for Call Park, on page 368
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**.

---

**What to do next**

Configure a Call Park Number, on page 369

---

**Configure a Call Park Number**

If you want to use Call Park across servers in a cluster, you must configure Call Park extension numbers on each server.

Ensure that each Call Park directory number, partition, and range is unique within the Cisco Unified Communications Manager. Each Cisco Unified Communications Manager to which devices are registered requires its own unique Call Park directory number and range. Cisco Unified Communications Manager Administration does not validate the Call Park numbers or ranges that you use to configure Call Park. To help identify invalid numbers or ranges and potential range overlaps, use the Cisco Unified Communications Manager Dialed Number Analyzer tool.

**Before you begin**

Configure a Partition for Call Park, on page 368

**Procedure**

**Step 1** Choose **Call Routing > Call Park**.

**Step 2** Perform one of the following tasks:
To add a new Call Park number, click Add New.

To copy a Call Park number, find the Call Park number or range of numbers and then click the Copy icon.

To update a Call Park number, find the Call Park number or range of numbers.

The Call Park number configuration window displays.

**Step 3** Configure the fields in the Call Park configuration fields. See the Related Topics section for more information about the fields and their configuration options.

**Step 4** To save the new or changed Call Park numbers in the database, click Save.

---

**What to do next**

Configure a Softkey Template for Call Park, on page 372

**Related Topics**

Call Park Configuration Fields, on page 371
## Call Park Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Call Park Number/Range        | Enter the Call Park extension number. You can enter digits or the wildcard character X (the system allows one or two Xs). For example, enter 5555 to define a single Call Park extension number of 5555 or enter 55XX to define a range of Call Park extension numbers from 5500 to 5599.  
  **Note** You can create a maximum of 100 Call Park numbers with one call park range definition. Make sure that the call park numbers are unique.  
  **Note** You cannot overlap call park numbers between Cisco Unified Communications Manager servers. Ensure that each Cisco Unified Communications Manager server has its own number range.  
  **Note** The call park range is selected from the list of servers where the call originates. For example, if phone A (registered to node A) calls phone B (registered to node B) and the phone B user presses Park, phone B requires a call park range in the CSS that resides on node A. In a multinode environment where phones and gateways communicate with various nodes and where calls that originate from any server may need to be parked, the phones require a CSS that contains call park ranges from all servers. |
| Description                   | Provide a brief description of this call park number. The description can include up to 50 characters in any language, but it cannot include double-quotes (“”), percentage sign (%), ampersand (&), or angle brackets (<>). |
| Partition                     | If you want to use a partition to restrict access to the call park numbers, choose the desired partition from the drop-down list. If you do not want to restrict access to the call park numbers, choose <None> for the partition.  
  **Note** Make sure that the combination of call park extension number and partition is unique within the Cisco Unified Communications Manager. |
Configure a Softkey Template for Call Park

Use this procedure to make the Park softkey available.

**Park** softkey has the following call states:
- On Hook
- Ring Out
- 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:

- If you modified a template that is already associated with devices, click **Apply Config** 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.

### 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**

**Step 1** Add a Softkey Template to a Common Device Configuration, on page 373

**Step 2** Associate a Common Device Configuration with a Phone, on page 374

### 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.

**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.

## Associate a Softkey 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 Call Park Button

### Configure a Phone Button Template for Call Park

### 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:

- If you modified a template that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

---

**What to do next**

*Associate a Button Template with a Phone, on page 237*

---

**Associate a Button Template with a Phone**

**Before you begin**

*Configure a Phone Button Template for Call Park, on page 374*

**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.

---

**What to do next**

(Optional) *Configure Park Monitoring, on page 376*
Configure Park Monitoring

Complete these optional tasks to add Park Monitoring to your Call Park configuration.

Before you begin
Park Monitoring is supported on only a subset of phones that support Call Park. The following Cisco Unified IP Phones support Park Monitoring:

- Cisco IP Phone 8811
- Cisco IP Phone 8841
- Cisco IP Phone 8845
- Cisco IP Phone 8851
- Cisco IP Phone 8851NR
- Cisco IP Phone 8861
- Cisco IP Phone 8865
- Cisco IP Phone 8865NR
- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Park Monitoring System Timers, on page 376</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Park Monitoring for Hunt Pilots, on page 377</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure Park Monitoring for a Directory Number, on page 378</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure Park Monitoring via Universal Line Template, on page 379</td>
</tr>
</tbody>
</table>

Configure Park Monitoring System Timers

Use this procedure to configure system-level timers for the Park Monitoring feature.
Procedure

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

Step 2 From the Server drop-down list box, select the publisher node.

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

Step 4 Configure values for the following service parameters:

- **Park Monitoring Reversion Timer**—The number of seconds that Cisco Unified Communications Manager waits before prompting the user to retrieve a parked call. For individual phone lines, this setting can be overridden by the same setting in the Directory Number Configuration window.

- **Park Monitoring Periodic Reversion Timer**—The number of seconds between reversion attempts when a call has been parked. Cisco Unified Communications Manager prompts the user about the parked call by ringing, beeping, or flashing the parker's phone.

- **Park Monitoring Forward No Retrieve Timer**—The number of seconds that park reminder notifications occur before the parked call is forwarded to the Park Monitoring Forward No Retrieve destination specified in the call parker's Directory Number configuration.

**Note** For additional details on these fields, see the service parameter online help.

Step 5 Click Save.

What to do next

Use any of these optional tasks to assign how expired timers get handled for individual phones lines and hunt pilots:

- Configure Park Monitoring for Hunt Pilots, on page 377
- Configure Park Monitoring for a Directory Number, on page 378
- Configure Park Monitoring via Universal Line Template, on page 379

Configure Park Monitoring for Hunt Pilots

If your deployment uses hunt pilots, use this optional procedure to assign a Park Monitoring destination to a hunt pilot.

**Note** For general information on setting up hunt pilots, see the "Configure Hunt Pilots" chapter of the System Configuration Guide for Cisco Unified Communications Manager.

Before you begin

Configure Park Monitoring System Timers, on page 376
Configure Park Monitoring for a Directory Number

Use this procedure to assign a Park Monitoring destination for an individual phone line. You can forward calls to another number, send to voicemail, or return to the call Parker.

**Note**
The following tools are available to provision settings for multiple phone lines:

- Use a universal line template to provision park monitoring settings for multiple phone lines via an LDAP directory sync. For details, see Configure Park Monitoring via Universal Line Template, on page 379.

- Use the Bulk Administration Tool to import a CSV file with settings for a large number of phone lines. For details, see the Bulk Administration Guide for Cisco Unified Communications Manager.

**Before you begin**
Configure Park Monitoring System Timers, on page 376

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Call Routing > Directory Number.

**Step 2**
Click Find and select the directory number that you want to configure.

**Step 3**
Enter values for the following Park Monitoring fields:

- **Park Monitoring Forward No Retrieve Destination External**—When the Park Monitoring Forward No Retrieve Timer expires, and the parkee is an external party, the call is forwarded either to voicemail or to a specified directory number. If this field is empty, the call is redirected to the call Parker’s line.

- **Park Monitoring Forward No Retrieve Destination Internal**—When the Park Monitoring Forward No Retrieve Timer expires, and the parkee is an internal party, the call is forwarded either to voicemail or to a specified directory number. If this field is empty, the call is redirected to the call Parker’s line.
• **Park Monitor Reversion Timer**—The number of seconds that Cisco Unified Communications Manager waits before prompting the user to retrieve a call parked on this phone line. If the value is 0 or empty, then Cisco Unified Communications Manager uses the value of the **Park Monitor Reversion Timer** service parameter.

**Step 4**
Complete any remaining fields in the **Directory Number Configuration** window. For help with the fields and their settings, refer to the online help.

**Step 5**
Click **Save**.

---

**Configure Park Monitoring via Universal Line Template**

Use this procedure to assign park monitoring settings to a universal line template. If you have an LDAP directory sync configured, you can use the universal line template configuration to provision directory number settings with park monitoring configured for multiple users.

**Before you begin**

Configure Park Monitoring System Timers, on page 376

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **User Management > User Phone/Add > Universal Line Template**.

**Step 2**
Perform one of the following steps:

- Click **Find** and select an existing template.
- Click **Add New** to create a new template.

**Step 3**
Expand the **Park Monitoring Settings** section and complete the fields. For field descriptions, see Park Monitoring Settings for Universal Line Templates, on page 379.

**Step 4**
Click **Save**.

---

**What to do next**

To apply the universal line template to individual directory numbers, you must assign the template to a user profile, feature group template, and LDAP directory sync. When the sync occurs, the template settings get applied to the phone lines that are a part of the sync. For LDAP setup, see the "Configure End Users" chapters in the **System Configuration Guide for Cisco Unified Communications Manager**.

**Park Monitoring Settings for Universal Line Templates**

The following table contains the Park Monitoring fields in the **Universal Line Template Configuration** window of Cisco Unified Communications Manager.
### Table 26: Park Monitoring Settings for Universal Line Templates

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Forward Destination for External Calls When Not Retrieved | When the person whose call is parked is an external party and the **Park Monitoring Forward No Retrieve Timer** expires, the system sends the call to one of these destinations:  
  - **Voicemail**—Uses the configuration in Voice Mail Profile to determine where to send the call.  
  - **Revert to Originator**—Returns the call to the call parker.  
  - To forward calls to another number, input the other number in the text box.  
  If no option is selected, the call returns to the call parker. |
| Calling Search Space for Forwarding External Calls When Not Retrieved | If you have configured parked calls to be redirected to a configured number, select the calling search space for the forward destination.  |
| Forward Destination for Internal Calls When Not Retrieved | When the person whose call is parked is an internal party and the **Park Monitoring Forward No Retrieve Timer** expires, the system sends the call to one of these destinations:  
  - **Voicemail**—Uses the configuration in Voice Mail Profile to determine where to send the call.  
  - **Revert to Originator**—Returns the call to the call parker.  
  - To forward calls to another number, input the other number in the text box.  
  If no option is selected, the call returns to the call parker. |
| Calling Search Space for Forwarding Internal Calls When Not Retrieved | If you have configured parked calls to be redirected to a configured number, select the calling search space for the forward destination.  |
| Park Monitor Reversion Timer (seconds)   | This timer determines the number of seconds that Unified Communications Manager waits before prompting the user to retrieve a call that the user parked. This timer starts when the user presses the Park softkey on the phone, and a reminder is issued when the timer expires. The default value is 60 seconds.  
  **Note** If you select 0 for the timer then phone lines that use this template will use the value of the **Park Monitor Reversion Timer** cluster-wide service parameter. |
Call Park Interactions and Restrictions

Call Park Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>CTI Applications</td>
<td>CTI applications access call park functionality, including monitoring activity on call park DNs. To monitor a call park DN, add an application or end user that is associated with the CTI application to the Standard CTI Allow Call Park Monitoring user group.</td>
</tr>
<tr>
<td>Music On Hold</td>
<td>Music On Hold allows users to place calls on hold with music that a streaming source provides. The Music On Hold audio source for Call Park is selected by the setting of the Network Hold MOH Audio Source setting within the Phone Configuration window. If you do not choose an audio source within the device configuration, Cisco Unified CM uses the audio source that is defined in the device pool or the system default if the device pool does not specify an audio source ID.</td>
</tr>
<tr>
<td>Route Plan Report</td>
<td>The route plan report displays the patterns and directory numbers that are configured in Cisco Unified Communications Manager. Use the route plan report to look for overlapping patterns and directory numbers before assigning a directory number to Call Park.</td>
</tr>
<tr>
<td>Calling Search Space and Partitions</td>
<td>Assign the call park directory number or range to a partition to limit call park access to users on the basis of the device calling search space.</td>
</tr>
<tr>
<td>Immediate Divert</td>
<td>Call Park supports Immediate Divert (iDivert or Divert softkey). For example, user A calls user B, and user B parks the call. User B retrieves the call and then decides to send the call to a voice-messaging mailbox by pressing the iDivert or Divert softkey. User A receives the voice mail greeting of user B.</td>
</tr>
</tbody>
</table>
| Barge                             | • Barge with Call Park–The target phone (the phone that is being barged upon) controls the call. The barge initiator “piggybacks” on the target phone. The target phone includes most of the common features, even when the target is being barged; therefore, the barge initiator has no feature access. When the target parks a call, the barge initiator then must release its call (the barge).  
  • cBarge with Call Park–The target and barge initiator act as peers. The cBarge feature uses a conference bridge, which causes it to function like a MeetMe conference. Both phones (target and barge initiator) have full access to their features. |
| Directed Call Park                | We recommend that you do not configure both Directed Call Park and the Park softkey for Call Park, but the possibility exists to configure both. If you configure both, ensure that the call park and directed call park numbers do not overlap. |
When a user parks a call across a QSIG intercluster trunk or a QSIG gateway trunk, the caller who has been parked (the parkee) does not see the To parked number message. The phone continues to display the original connected number. The call has been parked, and the user who parked the call can retrieve it. When the call is retrieved from the parked state, the call continues, but the caller who was parked does not see the newly connected number.

### Call Park Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>QSIG Intercluster Trunks</td>
<td>When a user parks a call across a QSIG intercluster trunk or a QSIG gateway trunk, the caller who has been parked (the parkee) does not see the To parked number message. The phone continues to display the original connected number. The call has been parked, and the user who parked the call can retrieve it. When the call is retrieved from the parked state, the call continues, but the caller who was parked does not see the newly connected number.</td>
</tr>
</tbody>
</table>

### Call Park Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park</td>
<td>Cisco Unified Communications Manager can park only one call at each call park extension number.</td>
</tr>
<tr>
<td>Shared Line</td>
<td>For shared line devices across nodes, the line registers to the node on which the device registers first. For example, if a device from subscriber 2 registers first and the line is created in subscriber 2 and the publisher node, the line belongs to subscriber 2. Each node must be configured with the call park number.</td>
</tr>
<tr>
<td>Backup</td>
<td>To achieve failover or fallback, configure call park numbers on the publisher node and subscriber nodes. With this configuration, when the primary node is down, the line device association gets changed to the secondary node, and the secondary node call park number gets used.</td>
</tr>
<tr>
<td>Directed Call Park</td>
<td>If a directed call park (or call park) is initiated from a shared line and the call is not retrieved from any device, the parked call does not always get reverted to the recipient in the shared line (parker).</td>
</tr>
<tr>
<td>Conference</td>
<td>When a conference call is set up between both the shared line and the caller on park reversion or park reversion fails causing a two-party call (between the other shared line and caller). The reason is that, on park reversion, Cisco Unified Communications Manager extends the call to both devices sharing the line and tries to add either party in conference (party already in conference or party that hit the park). If the party attempts to add the party who is already in the conference first, then the park reversion fails. When park reversion fails, the shared line can still barge into the call as usual.</td>
</tr>
<tr>
<td>Delete Server</td>
<td>If any call park numbers are configured for Cisco Unified Communications Manager on a node that is being deleted in the Server Configuration window (System &gt; Server), the node deletion fails. Before you can delete the node, you must delete the call park numbers in Cisco Unified Communications Manager Administration.</td>
</tr>
</tbody>
</table>
Troubleshooting Call Park

User Cannot Park Calls

**Problem**

User cannot park calls. When the user presses the Park softkey or feature button, the call does not get parked.

**Solution**

Ensure that a unique call park number is assigned to each Cisco Unified Communications Manager in the cluster.

The partition that is assigned to the call park number does not match the partition that is assigned to the phone directory number. For more information on partition see the *System Configuration Guide for Cisco Unified Communications Manager*.

Call Park Number is Not Displayed Long Enough

**Problem**

The call park number is not displayed long enough for the user.

**Solution**

Set the Call Park Display Timer to a longer duration. See the Related Topics section for more information about the Timer.

**Related Topics**

Configure Clusterwide Call Park, on page 367

Directed Call Park Overview

Directed Call Park allows a user to transfer a call to an available user-selected directed call park number. Configured Directed Call Park numbers exist cluster-wide. You can configure phones that support the directed call park Busy Lamp Field (BLF) to monitor the busy or idle status of specific directed call park numbers. Users can also use the BLF to speed dial a directed call park number.

Cisco Unified Communications Manager can park only one call at each directed call park number. To retrieve a parked call, a user must dial a configured retrieval prefix followed by the directed call park number at which the call is parked.
Directed Call Park Prerequisites

Make sure that the phones in your deployment support Directed Call Park. For a list of supported phones, run the Phone Feature List report from Cisco Unified Reporting, selecting Assisted Directed Call Park as the feature. For details, see Generate a Phone Feature List, on page 1.

Directed Call Park Configuration Task Flow

Before you begin

• Review Directed Call Park Prerequisites, on page 384

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure ClusterWide Directed Call Park, on page 384</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure a Directed Call Park Number, on page 385</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure BLF/Directed Call Park Buttons, on page 386</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Synchronize Directed Call Park with Affected Devices, on page 388</td>
</tr>
</tbody>
</table>

Configure ClusterWide Directed Call Park

Procedure

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

**Step 2**
To set the timer, update the Call Park Reversion Timer fields in the Clusterwide Parameter(Feature- General) section.

The default is 60 seconds. This parameter determines the time that a call remains parked. When this timer expires, the parked call returns to the device that parked the call or to another specified number, depending on what you configure in the Directed Call Park Configuration window.

What to do next
Configure a Directed Call Park Number, on page 385
Configure a Directed Call Park Number

Before you begin

Ensure that each directed call park directory number, partition, and range is unique within the Cisco Unified Communications Manager. Before you begin, generate a route plan report. If the Park softkey is also activated (not recommended), ensure that no overlap exists between call park numbers and directed call park numbers. If reversion number is not configured, the call reverts to the Parker (parking party) after the Call Park Reversion Timer expires.

Configure ClusterWide Directed Call Park, on page 384

Procedure

**Step 1** Choose Call Routing > Directed Call Park.

**Step 2** Perform one of the following tasks:

- To add a new directed call park number, click Add New.
- To copy a directed call park number, find the directed call park number or range of numbers and then click the Copy icon.
- To update a directed call park number, find the directed call park number or range of numbers.

The directed call park number configuration window is displayed.

**Step 3** Configure the fields in the Directed Call Park settings area. See the Related Topics section for more information about the fields and their configuration options.

**Step 4** To save the new or changed call park numbers in the database, click Save.

If you update a directed call park number, Cisco Unified Communications Manager reverts any call that is parked on that number only after the Call Park Reversion Timer expires.

**Step 5** Click Apply Config.

The Apply Configuration Information dialog is displayed.

**Step 6** Click OK.

**Step 7** If you are using BLF to monitor directed Call Park numbers, click Restart Devices button on the Directed Call Park Configuration window. This step is optional if you are using change notification.

What to do next

Configure BLF/Directed Call Park Buttons, on page 386

Related Topics

- Directed Call Park Configuration Settings, on page 386
### Directed Call Park Configuration Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number</td>
<td>Enter the directed call park number. You can enter digits (0-9) or the wildcard character ([], -, *, ^, #) and X (the system allows one or two Xs). For example, enter 5555 to define a single call park number of 5555 or enter 55XX to define a range of directed call park extension numbers from 5500 to 5599. Make sure that the directed call park numbers are unique and that they do not overlap with call park numbers.</td>
</tr>
<tr>
<td>Description</td>
<td>Provide a brief description of this directed call park number or range. The description can include up to 50 characters in any language, but it cannot include double quotation marks (&quot;), percentage sign (%), ampersand (&amp;), or angle brackets (&lt;&gt;) and tabs.</td>
</tr>
<tr>
<td>Partition</td>
<td>If you want to use a partition to restrict access to the directed call park numbers, choose the desired partition from the drop-down list. If you do not want to restrict access to the directed call park numbers, leave the partition as the default of &lt;None&gt;. <strong>Note</strong> Make sure that the combination of directed call park number and partition is unique within Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Reversion Number</td>
<td>Enter the number to which you want the parked call to return if not retrieved, or leave the field blank. <strong>Note</strong> A reversion number can comprise digits only; you cannot use wildcards.</td>
</tr>
<tr>
<td>Reversion Calling Search Space</td>
<td>Using 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>Retrieval Prefix</td>
<td>For this required field, enter the prefix for retrieving a parked call. The system needs the retrieval prefix to distinguish between an attempt to retrieve a parked call and an attempt to initiate a directed park.</td>
</tr>
</tbody>
</table>

### Configure BLF/Directed Call Park Buttons

**Before you begin**

Configure ClusterWide Directed Call Park, on page 384
Procedure

Step 1  From Cisco Unified Communications Manager Administration, choose Device > Device Settings > Phone Button Template. The Phone Button Template Configuration window displays.

Step 2  After the configuration window displays, click the Add a new BLF Directed Call Park link in the Association Information pane.

Note  The link does not display in the Association Information pane if the phone button template that you applied to the phone or device profile does not support BLF/Directed Call Park.

Step 3  Configure the fields in the BLF/Directed Call Park fields area. See the Related Topics section for more information about the fields and their configuration options.

Step 4  After you complete the configuration, click Save and close the window. The directory numbers are displayed in the Association Information pane of the Phone Configuration Window.

What to do next

Synchronize Directed Call Park with Affected Devices, on page 388

Related Topics

BLF/Directed Call Park Configuration Fields, on page 387

BLF/Directed Call Park Configuration Fields

Table 27: BLF/Directed Call Park Button Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory Number</td>
<td>The Directory Number drop-down list displays a list of Directed Call Park number that exist in the Cisco Unified Communications Manager database. For phones that are running SCCP or phones that are running SIP, choose the number (and corresponding partition, if it is displayed) that you want the system to dial when the user presses the speed-dial button; for example, 6002 in 3. Directory numbers that display without specific partitions belong to the default partition.</td>
</tr>
<tr>
<td>Label</td>
<td>Enter the text that you want to display for the BLF/Directed Call Park button. This field supports internationalization. If your phone does not support internationalization, the system uses the text that displays in the Label ASCII field.</td>
</tr>
</tbody>
</table>
Synchronize Directed Call Park with Affected Devices

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Choose Call Routing &gt; Directed Call Park. The Find and List Directed Call Parks window is displayed.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Choose the search criteria to use.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Find. The window displays a list of directed call parks that match the search criteria.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click the directed call park to which you want to synchronize applicable devices. The Directed Call Park Configuration window is displayed.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Make any additional configuration changes.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click Apply Config. The Apply Configuration Information dialog is displayed.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Click OK.</td>
</tr>
</tbody>
</table>

Directed Call Park Interactions and Restrictions

**Directed Call Park Interactions**

The following table describes feature interactions with the Directed Call Park feature.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Label ASCII</td>
<td>Enter the text that you want to display for the BLF/Directed Call Park button. The ASCII label represents the noninternationalized version of the text that you enter in the Label field. If the phone does not support internationalization, the system uses the text that displays in this field.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>If you enter text in the Label ASCII field that differs from the text in the Label field, Cisco Unified Communications Manager Administration accepts the configuration for both fields, even though the text differs.</td>
</tr>
</tbody>
</table>
**Interaction**

Music On Hold

The Music On Hold Audio Source for directed call park is assigned via the **Default Network Hold MOH Audio Source** service parameter. To assign the parameter:

1. From Cisco Unified CM Administration, choose **System > Service Parameters**.
2. From the **Server** drop-down, choose a Unified Communications Manger cluster node.
3. From the **Service** drop-down, select **Cisco CallManager**.
4. Under **Clusterwide Paramters (Service)**, assign a MOH audio source to the **Default Network Hold MOH Audio Source ID** parameter. The default is 1.
5. Click **Save**.

**Note**

For detailed information on adding MOH audio sources to the system, refer to the "Configure Music On Hold" section of this guide.

Calling Search Space and Partitions

Assign the Directed Call Park Directory number or range to a partition to limit Directed Call Park access to users on the basis of the device calling search space.

Immediate Divert

Directed call park supports Immediate Divert (iDivert or Divert softkey). For example, user A calls user B, and user B parks the call. User B retrieves the call and then decides to send the call to a voice-messaging mailbox by pressing the iDivert or Divert softkey. User A receives the voicemail greeting of user B.

Barge

- Barge with Directed Call Park–The target phone (the phone that is being barged upon) controls the call. The barge initiator “piggybacks” on the target phone. The target phone includes most of the common features, even when the target is being barged; therefore, the barge initiator has no feature access. When the target parks a call by using directed call park, the barge initiator then must release its call (the barge).
- cBarge with Directed Call Park–The target and barge initiator act as peers. The cBarge feature uses a conference bridge that makes it behave like a Meet-me conference. Both phones (target and barge initiator) retain full access to their features.

Call Park

We recommend that you do not configure both directed call park and the Park softkey for call park, but the possibility exists to configure both. If you configure both, ensure that the call park and directed call park numbers do not overlap.

A caller who has been parked (the parkee) by using the directed call park feature cannot, while parked, use the standard call park feature.
# Directed Call Park Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directed Call Park number</td>
<td>Cisco Unified Communications Manager one party can park only one call at each Directed Call Park number.</td>
</tr>
<tr>
<td></td>
<td>You cannot delete a Directed Call Park number that a device is configured to monitor (using the BLF button). A message indicates that the Directed Call Park number or range cannot be deleted because it is in use. To determine which devices are using the number, click the Dependency Records link on the Directed Call Park Configuration window.</td>
</tr>
<tr>
<td>Standard Call Park feature</td>
<td>A caller who has been parked (the parkee) by using the Directed Call Park feature cannot, while parked, use the standard call park feature.</td>
</tr>
<tr>
<td>Directed Call Park BLF</td>
<td>The Directed Call Park BLF cannot monitor a range of Directed Call Park numbers. A user can monitor only individual Directed Call Park numbers by using the Directed Call Park BLF. For example, if you configure a Directed Call Park number range 8X, you cannot use the Directed Call Park BLF to monitor that whole range of 80 to 89.</td>
</tr>
<tr>
<td>Directed Call Park for phones that are running SIP</td>
<td>The following limitations apply to Directed Call Park for phones that are running SIP:</td>
</tr>
<tr>
<td></td>
<td>• Directed Call Park gets invoked by using the Transfer softkey on Cisco Unified IP Phones 7940 and 7960 that are running SIP.</td>
</tr>
<tr>
<td></td>
<td>• The system does not support directed call park when the Blind Transfer softkey is used on Cisco Unified IP Phones 7940 and 7960 that are running SIP.</td>
</tr>
<tr>
<td></td>
<td>• The system does not support directed call park BLF on Cisco Unified IP Phones 7940 and 7960 that are running SIP, and third-party phones that are running SIP.</td>
</tr>
</tbody>
</table>

## Troubleshooting Directed Call Park

### User Cannot Retrieve Parked Calls

User cannot retrieve parked calls. After dialing the directed call park number to retrieve a parked call, the user receives a busy tone, and the IP phone displays the message, “Park Slot Unavailable”.

Ensure that the user dials the retrieval prefix followed by the directed call park number.

### User Cannot Park Calls

User cannot park calls. After the Transfer softkey (or Transfer button if available) is pressed and the directed call park number is dialed, the call does not get parked.
Ensure that the partition that is assigned to the call park number matches the partition that is assigned to the phone directory number. Ensure that the partition and calling search space are configured correctly for the device. For more information about the partition, see the System Configuration Guide for Cisco Unified Communications Manager.

**User Receives a Reorder Tone After the Reversion Timer Expires**

User cannot park calls. The user receives a reorder tone after the reversion timer expires.

Ensure that the user presses the Transfer softkey (or Transfer button if available) before dialing the directed call park number, and then presses the Transfer softkey (or Transfer button) again or goes on hook after dialing the directed call park number. Because directed call park is a transfer function, the directed call park number cannot be dialed alone.

---

**Note**

You can complete the transfer only by going on hook rather than pressing the Transfer softkey (or Transfer button) a second time if the Transfer On-hook Enabled service parameter is set to True.

---

**User Receives a Reorder Tone or Announcement**

User cannot park calls. After pressing the Transfer softkey (or Transfer button if available) and dialing the directed call park number, the user receives a reorder tone or announcement.

Ensure that the dialed number is configured as a directed call park number.

**User Cannot Park a Call at a Number Within The Range**

After configuring a range of directed call park numbers, the user cannot park a call at a number within the range.

Review the syntax for entering a range of directed call park numbers. If incorrect syntax is used, the system may appear to configure the range when it actually does not.

**Parked Calls Revert Too Quickly**

Parked calls revert too quickly.

Set the Call Park Reversion Timer to a longer duration.

**Park Slot Unavailable**

User cannot park calls. After pressing the Transfer softkey (or Transfer button if available) and dialing the directed call park number, the user receives a busy tone, and the IP phone displays the message, “Park Slot Unavailable”.

Ensure that the dialed directed call park number is not already occupied by a parked call or park the call on a different directed call park number.
Parked Calls Do Not Revert to the Parked Call Number

Parked calls do not revert to the number that parked the call.

Check the configuration of the directed call park number to ensure that it is configured to revert to the number that parked the call rather than to a different directory number.

Number or Range Cannot Be Deleted Because It Is in Use

When an attempt is made to delete a directed call park number or range, a message displays that indicates that the number or range cannot be deleted because it is in use.

You cannot delete a directed call park number that a device is configured to monitor (by using the BLF button). To determine which devices are using the number, click the Dependency Records link in the Directed Call Park Configuration window.
CHAPTER 31

Extension Mobility

- Extension Mobility Overview, on page 393
- Extension Mobility Prerequisites, on page 393
- Extension Mobility Configuration Task Flow, on page 393
- Extension Mobility Interactions and Restrictions, on page 407
- Extension Mobility Troubleshooting, on page 410

Extension Mobility Overview

Cisco Extension Mobility allows users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones within your system. If you have a single phone that will be used by multiple workers, for example, you can configure extension mobility so that individual users can log in to the phone and access their settings without affecting settings on other user accounts.

Extension Mobility Prerequisites

- A TFTP server that is reachable.
- Extension mobility functionality extends to most Cisco Unified IP Phones. Check the phone documentation to verify that Cisco Extension Mobility is supported.

Extension Mobility Configuration Task Flow

Before you begin

- Review Extension Mobility Prerequisites, on page 393

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that support the extension mobility feature.</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>Activate Extension Mobility Services, on page 394</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the Cisco Extension Mobility Phone Service, on page 395</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Create an Extension Mobility Device Profile for Users, on page 396</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Associate a Device Profile to a User, on page 402</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>Subscribe to Extension Mobility, on page 402</td>
<td></td>
</tr>
<tr>
<td>Step 7</td>
<td>Configure the Change Credential IP Phone Service, on page 403</td>
<td></td>
</tr>
<tr>
<td>Step 8</td>
<td>(Optional) Configure Service Parameters for Extension Mobility, on page 404</td>
<td></td>
</tr>
</tbody>
</table>

### Activate Extension Mobility Services

**Before you begin**

Generate a Phone Feature List, on page 1

**Procedure**

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

**Step 2** From the **Server** drop-down list, choose the required node.

**Step 3** Activate the following services:

a) Cisco CallManager
b) Cisco Tftp
c) Cisco Extension Mobility
d) ILS Service
Configure the Cisco Extension Mobility Phone Service

Configure the extension mobility IP phone service to which users can later subscribe to access extension mobility.

**Before you begin**

Activate Extension Mobility Services, on page 394

**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 Device &gt; Device Settings &gt; Phone Services.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Service Name field, enter a name for the service.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Service URL field, enter the Service URL.</td>
</tr>
</tbody>
</table>

The format is http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#. IP Address is the IP address of the Cisco Unified Communications Manager where Cisco Extension Mobility is activated and running.

It can either be a IPv4 or a IPv6 address.

**Example:**

http://123.45.67.89:8080/emapp/EMAppServlet?device=#DEVICENAME#

**Example:**


This format allows a user to sign-in using User ID and PIN. You can configure more sign-in options for IP phone users who have subscribed to the extension mobility service. To configure more sign-in options, append the loginType parameter to the Service URL, in the following formats:

- **loginType-DN** enables users to sign-in using Primary Extension and PIN.

  The Service URL format is: http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=DN.

- **loginType-SP** enables users to sign-in using Self Service User ID and PIN.

  The Service URL format is: http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=SP.
Create an Extension Mobility Device Profile for Users

Configure an extension mobility device profile. This profile acts as a virtual device that maps onto a physical device when a user logs in to extension mobility. The physical device takes on the characteristics in this profile.

Before you begin
Configure the Cisco Extension Mobility Phone Service, on page 395

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 choose an existing device profile from the resulting list.
  - To add a new device profile, click Add New, choose an option from the Device Profile Type, and click Next. Then, choose a device protocol from the Device Protocol drop-down list and click Next.
Step 3 Configure the fields. See the Related Topics section for more information about the fields and their configuration options.
Step 4 Click Save.
Step 5 From the Association Information section, click Add a new DN.
Step 6 In the Directory Number field, enter the directory number and click Save.
Step 7 Click Reset and follow the prompts.

What to do next
Associate a Device Profile to a User, on page 402

Step 5 In the Service Type field, choose whether the service is provisioned to the Services, Directories, or Messages button.
Step 6 Click Save.
**Device Profile Fields for Extension Mobility**

**Table 28: Device Profile Settings**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product Type</td>
<td>This field displays the product type to which this device profile applies.</td>
</tr>
<tr>
<td>Product Type</td>
<td>This field displays the product type to which this device profile applies.</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>This field displays the device protocol to which this device profile applies.</td>
</tr>
<tr>
<td>Device Profile Name</td>
<td>Enter a unique name. This name can comprise up to 50 characters in length.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description of the device profile. For text, use anything that describes this particular user device profile.</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>To specify the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list box. If you do not choose an audio source, Cisco Unified Communications Manager uses the audio source that is defined in the device pool or the system default if the device pool does not specify an audio source ID. <strong>Note</strong> You define audio sources in the Music On Hold Audio Source Configuration window. For access, choose Media Resources &gt; Music On Hold Audio Source.</td>
</tr>
<tr>
<td>User Locale</td>
<td>From the drop-down list box, choose the locale that is associated with the phone user interface. The user locale identifies a set of detailed information, including language and font, to support users. Cisco Unified Communications Manager makes this field available only for phone models that support localization. <strong>Note</strong> If no user locale is specified, Cisco Unified Communications Manager uses the user locale that is associated with the device pool. <strong>Note</strong> If the users require information to display (on the phone) in any language other than English, verify that the locale installer is installed before configuring user locale. See the Cisco Unified Communications Manager Locale Installer documentation.</td>
</tr>
<tr>
<td>Phone Button Template</td>
<td>From the Phone Button Template drop-down list, choose a phone button template. <strong>Tip</strong> If you want to configure BLF/SpeedDials for the profile for presence monitoring, choose a phone button template that you configured for BLF/SpeedDials. After you save the configuration, the Add a New BLF SD link displays in the Association Information pane. For more information on BLF/SpeedDials, see the Cisco Unified Communications Manager Features and Services Guide.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Softkey Template</td>
<td>From the Softkey Template drop-down list box, choose the softkey template from the list that displays.</td>
</tr>
<tr>
<td>Privacy</td>
<td>From the Privacy drop-down list box, choose On for each phone on which you want privacy. For more information, see the <em>Cisco Unified Communications Manager Features and Services Guide</em>.</td>
</tr>
<tr>
<td>Single Button Barge</td>
<td>From the drop-down list box, choose from the following options:</td>
</tr>
<tr>
<td></td>
<td>• Off—This device does not allow users to use the Single Button Barge/cBarge feature.</td>
</tr>
<tr>
<td></td>
<td>• Barge—Choosing this option allows users to press the Single Button Barge shared-line button on the phone to barge into a call using Barge.</td>
</tr>
<tr>
<td></td>
<td>• Default—This device inherits the Single Button Barge/cBarge setting from the service parameter and device pool settings.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting.</td>
</tr>
<tr>
<td></td>
<td>For more information, see the <em>Cisco Unified Communications Manager Features and Services Guide</em>.</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>From the drop-down list box, choose from the following options:</td>
</tr>
<tr>
<td></td>
<td>• Off—This device does not allow users to use the Join Across Lines feature.</td>
</tr>
<tr>
<td></td>
<td>• On—This device allows users to join calls across multiple lines.</td>
</tr>
<tr>
<td></td>
<td>• Default—This device inherits the Join Across Lines setting from the service parameter and device pool settings.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting.</td>
</tr>
<tr>
<td></td>
<td>For more information, see the Cisco Unified Communications Manager System Guide.</td>
</tr>
<tr>
<td>Always Use Prime Line</td>
<td>From the drop-down list box, 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—Cisco Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco CallManager service.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message</td>
<td>From the drop-down list box, choose one of the following options:</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>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td>To configure call display restrictions and ignore any presentation restriction that is received for internal calls, check the “Ignore Presentation Indicators (internal calls only)” check box.</td>
</tr>
<tr>
<td></td>
<td><strong>Tip</strong> 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. For more information about call display restrictions, see the Cisco Unified Communications Manager Features and Services Guide.</td>
</tr>
<tr>
<td>Do Not Disturb</td>
<td>Check this check box to enable Do Not Disturb.</td>
</tr>
<tr>
<td>DND Option</td>
<td>When you enable DND on the phone, this parameter allows you to specify how the DND feature handles incoming calls:</td>
</tr>
<tr>
<td></td>
<td>• Call Reject—This 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>
<tr>
<td></td>
<td>• Ringer Off—This option turns off the ringer, but incoming call information gets presented to the device, so that the user can accept the call.</td>
</tr>
<tr>
<td></td>
<td>• Use Common Phone Profile Setting—This option specifies that the DND Option setting from the Common Phone Profile window will get used for this device.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> For 7940/7960 phones that are running SCCP, you can only choose the Ringer Off option. For mobile devices and dual-mode phones, you can only choose the Call Reject option. When you activate DND Call Reject on a mobile device or dual-mode phone, no call information gets presented to the device.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| DND Incoming Call Alert      | When you enable the DND Ringer Off or Call Reject option, this parameter specifies how a call displays on a phone. 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 will get 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 phone to play a beep tone only.  
  • Flash Only—For an incoming call, this option causes the phone to display a flash alert.                                                                                                                                 |
| Extension Mobility Cross Cluster CSS | From the drop-down list box, choose an existing Calling Search Space (CSS) to use for this device profile for the Extension Mobility Cross Cluster feature. (To configure a new CSS or modify an existing CSS, choose Call Routing > Class of Control > Calling Search Space in Cisco Unified Communications Manager Administration.)  
  Default value specifies None.  
  The home administrator specifies this CSS, which gets used as the device CSS that gets assigned to the phone when the user logs in to this remote phone. For more information, see the Cisco Unified Communications Manager Features and Services Guide.                                                                 |
| Module 1                     | You can configure one or two expansion modules for this device profile by choosing phone templates from the expansion module drop-down lists in the expansion module fields.  
  **Note** You can view a phone button list at any time by choosing the View button list link next to the phone button template fields. A separate dialog box pops up and displays the phone buttons for that particular expansion module.  
  Choose the appropriate expansion module or None.                                                                                                                                                                      |
| Module 2                     | Choose the appropriate expansion module or None.                                                                                                                                                           |
| MLPP Domain                  | If this user device profile will be used for MLPP precedence calls, choose the MLPP Domain from the drop-down list box.  
  **Note** You define MLPP domains in the MLPP Domain Configuration window. For access, choose System > MLPP Domain.                                                                                                  |
### Field | Description
--- | ---
MLPP Indication | If this user device profile will be used for MLPP precedence calls, assign an MLPP Indication setting to the device profile. 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 box, choose a setting to assign to this device profile from the following options:

1. Default—This device profile inherits its MLPP indication setting from the device pool of the associated device.
2. Off—This device does not handle nor process indication of an MLPP precedence call.
3. On—This device profile does handle and process indication of an MLPP precedence call.

**Note** | Do not configure a 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.

MLPP Preemption | If this user device profile will be used for MLPP precedence calls, assign an MLPP Preemption setting to the device profile. 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 box, choose a setting to assign to this device profile from the following options:

1. Default—This device profile inherits its MLPP preemption setting from the device pool of the associated device.
2. Disabled—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.
3. Forceful—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.

**Note** | Do not configure a device 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.

Login User Id | From the Login User ID drop-down list box, choose a valid login user ID.

**Note** | If the device profile is used as a logout profile, specify the login user ID that will be associated with the phone. After the user logs out from this user device profile, the phone will automatically log in to this login user ID.
Associate a Device Profile to a User

Associate a device profile to users so that they can access their settings from a different phone. You associate a user device profile to a user in the same way that you associate a physical device.

Tip
You can use the Bulk Administration Tool (BAT) to add and delete several user device profiles for Cisco Extension Mobility at one time. 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.

Before you begin
Create an Extension Mobility Device Profile for Users, on page 396

Procedure

Step 1
From Cisco Unified CM Administration, choose User Management > End User.

Step 2
Perform one of the following tasks:

• To modify the settings for an existing user, enter search criteria, click Find, and choosing an existing user from the resulting list.
• To add a new user, click Add New.

Step 3
Under Extension Mobility, locate the device profile that you created and move it from Available Profiles to Controlled Profiles.

Step 4
Check the Home Cluster check box.

Step 5
Click Save.

What to do next
Subscribe to Extension Mobility, on page 402

Subscribe to Extension Mobility

Subscribe IP phones and device profiles to the extension mobility service so that users can log in, use, and log out of extension mobility.

Before you begin
Associate a Device Profile to a User, on page 402

Procedure

Step 1
Perform one of the following tasks from Cisco Unified CM Administration:
Configure the Change Credential IP Phone Service

To allow users to change their PINs on their phones, you must configure the change credential Cisco Unified IP Phone service and associate the user, the device profile, or the IP phone with the change credential phone service.

**Before you begin**

Subscribe to Extension Mobility, on page 402

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Device > Device Settings > Phone Services.

**Step 2**
Click Add New.

**Step 3**
In the Service Name field, enter Change Credential.

**Step 4**
In the Service URL field, enter the following value, where server designates the server where the Change Credential IP phone service runs:

```
http://server:8080/changecredential/ChangeCredentialServlet?device=#DEVICENAME#
```

**Step 5**
(Optional) In the Secure-Service URL field, enter the following value, where server is the server where the Change Credential IP phone service runs:

```
https://server:8443/changecredential/ChangeCredentialServlet?device=#DEVICENAME#
```

**Step 6**
Configure the remaining fields in the IP Phone Services Configuration window, and choose Save.

**Step 7**
To subscribe the Cisco Unified IP Phone to the Change Credential IP phone service, choose Device > Phone.

**Step 8**
In the Phone Configuration window, go to the Related Links drop-down list and choose Subscribe/Unsubscribe Services.

**Step 9**
Click Go.

**Step 10**
From the Select a Service drop-down list, choose the Change Credential IP phone service.

**Step 11**
Click Next.

**Step 12**
Click Subscribe.

**Step 13**
Click Save.
What to do next
(Optional) Configure Service Parameters for Extension Mobility, on page 404

Configure Service Parameters for Extension Mobility

If you want to modify the behavior of extension mobility, configure the service parameters.

Before you begin
Activate Extension Mobility Services, on page 394

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Service Parameters.
Step 2 In the Server field, choose the node that is running the Cisco Extension Mobility service.
Step 3 In the Service field, choose Cisco Extension Mobility.
Step 4 Click Advanced to show all service parameters.
Step 5 Click Save.

See the Related Topics section for more information about these service parameters and their configuration options.

Extension Mobility Service Parameters

Table 29: Extension Mobility Service Parameters

<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enforce Intra-cluster Maximum Login Time</td>
<td>Select True to specify a maximum time for local logins. After this time, the system automatically logs out the device. False, which is the default setting, means that no maximum time for logins exists.</td>
</tr>
<tr>
<td></td>
<td>To set an automatic logout, you must choose True for this service parameter and also specify a system maximum login time for the Intra-cluster Maximum Login Time service parameter. Cisco Unified Communications Manager then uses the automatic logout service for all logins.</td>
</tr>
<tr>
<td></td>
<td>If the value of Enforce Intra-cluster Maximum Login Time is set to False and you specify a valid maximum login time for the Intra-cluster Maximum Login Time service parameter, then the value of Enforce Intra-cluster Maximum Login Time automatically changes to True.</td>
</tr>
</tbody>
</table>
## Service Parameter

<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intra-cluster Maximum Login Time</td>
<td>This parameter sets the maximum time that a user can be locally logged in to a device, such as 8:00 (8 hours) or :30 (30 minutes). The system ignores this parameter and set the maximum login time to 0:00, if the <strong>Enforce Intra-cluster Maximum Login Time</strong> parameter is set to <strong>False</strong>. Valid values are between 0:00 and 168:00 in the format HHH:MM, where HHH represents the number of hours and MM represents the number of minutes. <strong>Note</strong>: If you grant a user access to set their Extension Mobility maximum login time (configured via the <strong>Allow End User to set their Extension Mobility maximum login time</strong> check box in the <strong>User Profile Configuration</strong>) the user's configuration in the Self-Care Portal overrides the value of the <strong>Intra-cluster Maximum Login Time</strong> service parameter.</td>
</tr>
<tr>
<td>Maximum Concurrent Requests</td>
<td>Specify the maximum number of login or logout operations that can occur simultaneously. This number prevents the Cisco Extension Mobility service from consuming excessive system resources. The default value of 5 is acceptable in most cases.</td>
</tr>
</tbody>
</table>
| Multiple Login Behavior                | When users are logged in to one phone and then login to a second phone either in the same cluster or on a different cluster, users can view the login behavior on the second phone based on the **Multiple Login Behavior** setting defined on the **Service Parameter Configuration** page. Choose one of the following options from the drop–down list box:  
  - **Multiple Logins Allowed**—You can login to more than one device at a time.  
  - **Multiple Logins Not Allowed**—You can be logged in to only one device. The login attempts to the second device fails and the phone displays the error code “25” (Multi-Login Not Allowed). You can login successfully, only when you have logged out from the first device. This is the default value.  
  - **Auto Logout**—When you try to login to a second device (either Extension Mobility or Extension Mobility Cross Cluster), the Cisco Unified Communications Manager automatically logs you out of the first device.  
**Note**: Multiple login behavior is also applicable between two Extension Mobility Cross Cluster logins. |
| Alphanumeric User ID                   | Choose **True** to allow the user ID to contain alphanumeric characters. Choosing **False** allows the user ID to contain only numeric characters. **Note**: The Alphanumeric User ID parameter applies systemwide. You can have a mix of alphanumeric and numeric user IDs. The system supports only user IDs that can be entered by using the alphanumeric keypad. The case-sensitive userid field requires the characters to be lowercase. |
### Service Parameter | Description
--- | ---
**Remember the Last User Logged In** | When you choose **False**, the system does not remember the last user who logged in to the phone. Use this option when the user access the phone on a temporary basis only. Choose True to remember the last user that logged into the phone. Use this option when a phone has only one user.

For example, Cisco Extension Mobility is used to enable the types of calls that are allowed from a phone. Individuals who are not logged in and who are using their office phone can make only internal or emergency calls. But after logging in using Cisco Extension Mobility, the user can make local, long-distance, and international calls. In this scenario, only this user regularly logs in to the phone. It makes sense to set the Cisco Extension Mobility to remember the last user ID that logged in.

**Clear Call Logs on Intra-cluster EM** | Choose **True** to specify that the call logs are cleared during the Cisco Extension Mobility manual login and logout process.

While a user is using the Cisco Extension Mobility service on an IP phone, all calls (placed, received, or missed) appear in a call log and can be retrieved and seen on the IP phone display. To ensure privacy, set the Clear Call Log service parameter to **True**. This ensures that the call logs are cleared when a user logs out and another user logs in.

For extension mobility cross cluster (EMCC), the call log is always cleared when the user logs in or out of a phone.

**Note** | Call logs are cleared only during manual login/logout. If a Cisco Extension Mobility logout occurs automatically or any occurrence other than a manual logout, the call logs are not cleared.

**Validate IP Address** | This parameter sets whether validation occurs on the IP address of the source that is requesting login or logout.

If the parameter is set to **True**, the IP address from which a Cisco Extension Mobility log in or log out request occurs and is validated to ensure that it is trusted.

Validation is first performed against the cache for the device that will log in or log out.

If the IP address is found in the cache or in the list of trusted IP addresses or is a registered device, the device can log in or log out. If the IP address is not found, the log in or log out attempt is blocked.

If the parameter is set to **False**, the Cisco Extension Mobility log in or log out request is not validated.

Validation of IP addresses can affect the time that is required to log in or log out a device, but it offers additional security that prevents unauthorized log in or log out attempts. This function is recommended, especially when used with logins from separate trusted proxy servers for remote devices.

**Trusted List of IPs** | This parameter appears as a text box (the maximum length is 1024 characters). You can enter strings of trusted IP addresses or hostnames which are separated by semicolons, in the text box. IP address ranges and regular expressions are not supported.
### Service Parameter Description

<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allow Proxy</td>
<td>If the parameter is <strong>True</strong>, the Cisco Extension Mobility log in and log out operations that use a web proxy are allowed.</td>
</tr>
<tr>
<td></td>
<td>If the parameter is <strong>False</strong>, the Cisco Extension Mobility log in and log out requests coming from behind a proxy get rejected.</td>
</tr>
<tr>
<td></td>
<td>The setting that you select takes effect only if the Validate IP Address parameter specifies true.</td>
</tr>
<tr>
<td>Extension Mobility Cache Size</td>
<td>In this field, enter the size of the device cache that is maintained by Cisco Extension Mobility. The minimum value for this field is 1000 and the maximum is 20000. The default value is 10000.</td>
</tr>
<tr>
<td></td>
<td>The value that you enter takes effect only if the Validate IP Address parameter is <strong>True</strong>.</td>
</tr>
</tbody>
</table>

### Extension Mobility Interactions and Restrictions

#### Cisco Extension Mobility Interactions

**Table 30: Cisco Extension Mobility Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Assistant</td>
<td>A manager who uses Cisco Extension Mobility can simultaneously use Cisco Unified Communications Manager Assistant. The manager logs in to the Cisco Unified IP Phone by using Cisco Extension Mobility and then chooses the Cisco IP Manager Assistant service. When the Cisco IP Manager Assistant service starts, the manager can access assistants and all Cisco Unified Communications Manager Assistant features (such as call filtering and Do Not Disturb).</td>
</tr>
<tr>
<td>BLF Presence</td>
<td>When you configure BLF/speed dial buttons in a user device profile, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF/SpeedDial buttons after you log in to the device. When the extension mobility user logs out, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF/SpeedDial buttons for the logout profile that is configured.</td>
</tr>
<tr>
<td>Call Display Restrictions</td>
<td>When you enable call display restrictions, Cisco Extension Mobility functions as usual: when a user is logged in to the device, the presentation or restriction of the call information depends on the user device profile that is associated with that user. When the user logs out, the presentation or restriction of the call information depends on the configuration that is defined for that phone type in the Phone Configuration window. To use call display restrictions with Cisco Extension Mobility, check the <strong>Ignore Presentation Indicators (internal calls only)</strong> check box in both the Device Profile Configuration window and the Phone Configuration window.</td>
</tr>
</tbody>
</table>
### Call Forward All Calling Search Space

An enhancement to call forward all calling search space (CSS) lets you upgrade to later releases of Cisco Unified Communications Manager without loss of functionality.

The **CFA CSS Activation Policy** service parameter supports this enhancement. In the **Service Parameter Configuration** window, this parameter displays in the **Clusterwide Parameters (Feature - Forward)** section with two options:

- With Configured CSS (default)
- With Activating Device/Line CSS

### Do Not Disturb

For extension mobility, the device profile settings include do not disturb (DND) incoming call alert and DND status. When a user logs in and enables DND, the DND incoming call alert and DND status settings are saved, and these settings are used when the user logs in again.

**Note** When a user who is logged in to extension mobility modifies the DND incoming call alert or DND status settings, this action does not affect the actual device settings.

### Intercom

Cisco Extension Mobility supports the intercom feature. To support intercom, Cisco Extension Mobility uses a default device that is configured for an intercom line. An intercom line is presented on only the default device.

You can assign an intercom line to a device profile. When a user logs in to a device that is not the default device, the intercom line is not presented.

The following additional considerations apply to intercom for Cisco Extension Mobility:

- When Unified Communications Manager assigns an intercom line to a device and the default device value is empty, the current device is selected as the default device.
- When AXL programatically assigns an intercom DN, you must update the intercom DN separately by using Cisco Unified Communications Manager Administration to set the default device.
- When you delete a device that is set as the intercom default device for an intercom line, the intercom default device is no longer set to the deleted device.

### Internet Protocol Version 6 (IPv6)

Cisco Extension Mobility Supports IPv6. You can use phones with an IP addressing mode of IPv6 or dual-stack (IPv4 and IPv6).

### Prime Line

If you select **On** for the **Always Use Prime Line** parameter in the **Device Profile** or **Default Device Profile Configuration** window, a Cisco Extension Mobility user can use this feature after logging in to the device that supports Cisco Extension Mobility.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forward All Calling Search Space</td>
<td>An enhancement to call forward all calling search space (CSS) lets you upgrade to later releases of Cisco Unified Communications Manager without loss of functionality. The <strong>CFA CSS Activation Policy</strong> service parameter supports this enhancement. In the <strong>Service Parameter Configuration</strong> window, this parameter displays in the <strong>Clusterwide Parameters (Feature - Forward)</strong> section with two options:</td>
</tr>
<tr>
<td>Do Not Disturb</td>
<td>For extension mobility, the device profile settings include do not disturb (DND) incoming call alert and DND status. When a user logs in and enables DND, the DND incoming call alert and DND status settings are saved, and these settings are used when the user logs in again. <strong>Note</strong> When a user who is logged in to extension mobility modifies the DND incoming call alert or DND status settings, this action does not affect the actual device settings.</td>
</tr>
<tr>
<td>Intercom</td>
<td>Cisco Extension Mobility supports the intercom feature. To support intercom, Cisco Extension Mobility uses a default device that is configured for an intercom line. An intercom line is presented on only the default device. You can assign an intercom line to a device profile. When a user logs in to a device that is not the default device, the intercom line is not presented. The following additional considerations apply to intercom for Cisco Extension Mobility:</td>
</tr>
<tr>
<td>Prime Line</td>
<td>If you select <strong>On</strong> for the <strong>Always Use Prime Line</strong> parameter in the <strong>Device Profile</strong> or <strong>Default Device Profile Configuration</strong> window, a Cisco Extension Mobility user can use this feature after logging in to the device that supports Cisco Extension Mobility.</td>
</tr>
</tbody>
</table>
# Cisco Extension Mobility Restrictions

## Table 31: Cisco Extension Mobility Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cache</td>
<td>Cisco Extension Mobility maintains a cache of all logged-in user information for 2 minutes. If a request comes to extension mobility regarding a user who is represented in the cache, the user is validated with information from the cache. For example, if a user changes the password, logs out, and then logs back in within 2 minutes, both the old and new passwords are recognized.</td>
</tr>
<tr>
<td>Call Back</td>
<td>When a Cisco Extension Mobility user logs out of a device, all call back services that are active for the Cisco Extension Mobility user are automatically cancelled.</td>
</tr>
<tr>
<td>Character Display</td>
<td>The characters that display when a user logs in depend on the current locale of the phone. For example, if the phone is currently in the English locale (based on the Logout profile of the phone), the user can only enter English characters in the UserID.</td>
</tr>
<tr>
<td>Hold Reversion</td>
<td>Cisco Extension Mobility does not support the hold reversion feature.</td>
</tr>
<tr>
<td>IP Phones</td>
<td>Cisco Extension Mobility requires a physical Cisco Unified IP Phone for login. Users of office phones that are configured with Cisco Extension Mobility cannot remotely log in to their phones.</td>
</tr>
<tr>
<td>Locale</td>
<td>If the user locale that is associated with the user or profile is not the same as the locale or device, after a successful login, the phone will restart and then reset. This behavior occurs because the phone configuration file is rebuilt. Addon-module mismatches between profile and device can cause the same behavior.</td>
</tr>
<tr>
<td>Log Out</td>
<td>If Cisco Extension Mobility is stopped or restarted, the system does not automatically log out users who are already logged in after the logout interval expires. Those phones automatically log out users only once a day. You can manually log out these users from either the phones or from Cisco Unified CM Administration.</td>
</tr>
<tr>
<td>Secure Tone</td>
<td>Cisco Extension Mobility and join across line services are disabled on protected phones.</td>
</tr>
<tr>
<td>User Group</td>
<td>Although you can add users to the Standard EM authentication proxy rights user group, those users are not authorized to authenticate by proxy.</td>
</tr>
<tr>
<td>Remember the Last User Logged In</td>
<td>The service parameter <strong>Remember the Last User Logged In</strong> is applicable only for default Extension Mobility service URL or the Extension Mobility service URL with <code>loginType</code> as UID.</td>
</tr>
</tbody>
</table>
Troubleshoot Extension Mobility

Procedure

- Configure the Cisco Extension Mobility trace directory and enable debug tracing by performing the following steps:
  a) From Cisco Unified Serviceability, choose Trace > Trace Configuration.
  b) From the Servers drop-down list, select a server.
  c) From the Configured Services drop-down list, select Cisco Extension Mobility.
- Make sure that you entered the correct URL for the Cisco Extension Mobility service. Remember that the URL is case sensitive.
- Check that you have thoroughly and correctly performed all the configuration procedures.
- If a problem occurs with authentication of a Cisco Extension Mobility user, go to the user pages and verify the PIN.

What to do next

If you are still having problems, see the troubleshooting topics that follow.

Authentication Error

Problem “Error 201 Authentication Error” appears on the phone.

Solution The user should check that the correct user ID and PIN were entered; the user should check with the system administrator that the user ID and PIN are correct.

Blank User ID or PIN

Problem “Error 202 Blank User ID or PIN” appears on the phone.

Solution Enter a valid user ID and PIN.

Busy Please Try Again

Problem “Error 26 Busy Please Try Again” appears on the phone.

Solution Check whether the number of concurrent login and logout requests is greater than the Maximum Concurrent requests service parameter. If so, lower the number of concurrent requests.

Note To verify the number of concurrent login and logout requests, use the Cisco Unified Real-Time Monitoring Tool to view the Requests In Progress counter in the Extension Mobility object. For more information, see the Cisco Unified Real-Time Monitoring Tool Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html
Database Error

Problem “Error 6 Database Error” appears on the phone.

Solution Check whether a large number of requests exists. If a large number of requests exists, the Requests In Progress counter in the Extension Mobility object counter shows a high value. If the requests are rejected because of a large number of concurrent requests, the Requests Throttled counter also shows a high value. Collect detailed database logs.

Dev Logon Disabled

Problem “Error 22 Dev Logon Disabled” appears on the phone.

Solution Verify that you checked the Enable Extension Mobility check box in the Phone Configuration window (Device > Phone).

Device Name Empty

Problem “Error 207 Device Name Empty” appears on the phone.

Solution Check that the URL that is configured for Cisco Extension Mobility is correct. See the Related Topics section for more information.

Related Topics
- Configure the Cisco Extension Mobility Phone Service, on page 395

EM Service Connection Error

Problem “Error 207 EM Service Connection Error” appears on the phone.

Solution Verify that the Cisco Extension Mobility service is running by selecting Tools > Control Center—Feature in Cisco Unified Serviceability.

Host Not Found

Problem The “Host Not Found” error message appears on the phone.

Solution Check that the Cisco Tomcat service is running by selecting Tools > Control Center—Network Services in Cisco Unified Serviceability.

HTTP Error

Problem HTTP Error (503) appears on the phone.

Solution
- If you get this error when you press the Services button, check that the Cisco IP Phone Services service is running by selecting Tools > Control Center—Network Services in Cisco Unified Serviceability.
- If you get this error when you select Extension Mobility service, check that the Cisco Extension Mobility Application service is running by selecting Tools > Control Center—Network Services in Cisco Unified Serviceability.
Phone Resets

**Problem** After users log in or log out, their phones reset instead of restarting.

**Possible Cause** Locale change is the probable cause of the reset.

**Solution** No action is required. If the user locale that is associated with the logged-in user or profile is not the same as the locale or device, after a successful login the phone will restart and then reset. This pattern occurs because the phone configuration file is rebuilt.

Phone Services Unavailable After Login

**Problem** After logging in, the user finds that the phone services are not available.

**Possible Cause** This problem occurs because the user profile had no services associated with it when it was loaded on the phone.

**Solution**

- Ensure that the user profile includes the Cisco Extension Mobility service.
- Change the configuration of the phone where the user is logged in to include Cisco Extension Mobility. After the phone is updated, the user can access the phone services.

Phone Services Unavailable After Logout

**Problem** After a user logs out and the phone reverts to the default device profile, the phone services are no longer available.

**Solution**

- Verify that the *Synchronization Between Auto Device Profile and Phone Configuration* enterprise parameter is set to *True*.
- Subscribe the phone to the Cisco Extension Mobility service.

User Logged in Elsewhere

**Problem** “Error 25 User Logged in Elsewhere” appears on the phone.

**Solution** Check whether the user is logged in to another phone. If multiple logins must be allowed, ensure that the *Multiple Login Behavior* service parameter is set to *Multiple Logins Allowed*.

User Profile Absent

**Problem** “Error 205 User Profile Absent” appears on the phone.

**Solution** Associate a device profile to the user.
Extension Mobility Cross Cluster Overview

The extension mobility cross cluster (EMCC) feature provides users with the same functionality as extension mobility, but also allows them to move from one cluster (the home cluster) and log in to a temporary phone on another remote cluster (the visiting cluster). From there, they can access their phone settings from any location as if they were using an IP phone at the home office.

Extension Mobility Cross Cluster Prerequisites

- Other call-control entities that support and use the extension mobility cross cluster (EMCC) configuration; for example, other Cisco Unified Communications Manager clusters, EMCC intercluster service profiles, and EMCC remote cluster services
- Clusters that are set to nonsecure or mixed mode. See Extension Mobility Cross Cluster and Security Mode for Different Cluster Versions, on page 440 for more information.
- Supported phones in secure or nonsecure mode

Extension Mobility Cross Cluster Configuration Task Flow

Before you begin

- Review Extension Mobility Cross Cluster Prerequisites, on page 413
- Review Extension Mobility Cross Cluster Interactions and Restrictions, on page 436
### 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 1</td>
<td>Generate a report to identify devices that support the extension mobility cross cluster feature.</td>
</tr>
<tr>
<td><strong>Step 2</strong> To Configure Extension Mobility, on page 415, perform the following subtasks:</td>
<td>Configure extension mobility to allow users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones in one cluster. Perform this task flow on both home and remote clusters, so that users will be able to access settings from either a home or visiting cluster.</td>
</tr>
<tr>
<td>• Activate Services for Extension Mobility Cross Cluster, on page 416</td>
<td></td>
</tr>
<tr>
<td>• Configure the Extension Mobility Phone Service, on page 416</td>
<td></td>
</tr>
<tr>
<td>• Configure a Device Profile for Extension Mobility Cross Cluster, on page 417</td>
<td></td>
</tr>
<tr>
<td>• Enable Extension Mobility Cross Cluster for a User, on page 423</td>
<td></td>
</tr>
<tr>
<td>• Subscribe Devices to Extension Mobility, on page 424</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> To Configure Certificates for Extension Mobility Cross Cluster, on page 424, perform the following subtasks:</td>
<td>To configure the home and remote clusters properly, you must export certificates on each cluster to the same SFTP server and SFTP directory and consolidate them on one of the participating clusters. This procedure ensures that trust is established between the two clusters.</td>
</tr>
<tr>
<td>• Activate the Bulk Provisioning Service, on page 425</td>
<td></td>
</tr>
<tr>
<td>• Configure Bulk Certificate Management and Export Certificates, on page 425</td>
<td></td>
</tr>
<tr>
<td>• Consolidate the Certificates, on page 426</td>
<td></td>
</tr>
<tr>
<td>• Import the Certificates in to the Clusters, on page 427</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> To Configure Extension Mobility Cross Cluster Devices and Templates, on page 428, perform the following subtasks:</td>
<td>Configure a geolocation filter to specify criteria for device location matching, such as country, state, and city values. Geolocations are used to identify the location of a device, and the filter indicates what parts of the geolocation are significant.</td>
</tr>
<tr>
<td>• Create a Common Device Configuration, on page 429</td>
<td></td>
</tr>
<tr>
<td>• Configure an Extension Mobility Cross Cluster Template, on page 429</td>
<td></td>
</tr>
<tr>
<td>• Set the Default Template, on page 430</td>
<td></td>
</tr>
<tr>
<td>• Add Extension Mobility Cross Cluster Devices, on page 430</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure a Geolocation Filter for Extension Mobility Cross Cluster, on page 431</td>
<td></td>
</tr>
</tbody>
</table>
Configure Extension Mobility

Configure extension mobility to allow users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones in one cluster. Perform this task flow on both home and remote clusters, so that users will be able to access settings from either a home or visiting cluster.

Before you begin

Generate a Phone Feature List, on page 1

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 Services for Extension Mobility Cross Cluster, on page 416</td>
<td>Create the Extension Mobility phone service to which you can subscribe your users.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure the Extension Mobility Phone Service, on page 416</td>
<td>Create a device profile to map settings onto a real device when a user logs in to Extension Mobility cross cluster.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Device Profile for Extension Mobility Cross Cluster, on page 417</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Enable Extension Mobility Cross Cluster for a User, on page 423</td>
<td>Enable Extension Mobility on devices and subscribe to the service if you have not set up an enterprise subscription for all devices.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Subscribe Devices to Extension Mobility, on page 424</td>
<td></td>
</tr>
</tbody>
</table>
Activate Services for Extension Mobility Cross Cluster

**Before you begin**

Generate a Phone Feature List, on page 1

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified Serviceability, choose Tools &gt; Service Activation.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list, choose the required node.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Activate the following services:</td>
</tr>
<tr>
<td></td>
<td>a) Cisco CallManager</td>
</tr>
<tr>
<td></td>
<td>b) Cisco Tftp</td>
</tr>
<tr>
<td></td>
<td>c) Cisco Extension Mobility</td>
</tr>
<tr>
<td></td>
<td>d) ILS Service</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> You must choose publisher node to activate the ILS services.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 4</th>
<th>Click Save.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 5</td>
<td>Click OK.</td>
</tr>
</tbody>
</table>

**What to do next**

Configure the Extension Mobility Phone Service, on page 416

Configure the Extension Mobility Phone Service

Create the Extension Mobility phone service to which you can subscribe your users.

**Before you begin**

Activate Services for Extension Mobility Cross Cluster, on page 416

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; Phone Services.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Service Name field, enter a name for the service.</td>
</tr>
<tr>
<td></td>
<td>For example, enter a name such as Extension Mobility or EM. For Java MIDlet services,</td>
</tr>
<tr>
<td></td>
<td>the service name must exactly match the name that is defined in the Java Application</td>
</tr>
<tr>
<td></td>
<td>Descriptor (JAD) file.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Service URL field, enter the service URL in the following format:</td>
</tr>
<tr>
<td></td>
<td>http://&lt;IP Address&gt;:8080/emapp/EMAppServlet?device=#DEVICENAME#&amp;EMCC=#EMCC#.</td>
</tr>
</tbody>
</table>
Step 5  (Optional) If you want to create a secure URL using HTTPS, enter the secure service URL in the following format:
https://<IP Address>:8443/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#

Step 6  (Optional) If you want to configure more sign-in options, append the loginType parameter to the Service URL in the following formats:
- loginType=DN enables users to sign in using Primary Extension and PIN. The Service URL format is:
  http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#&loginType=DN.
- loginType=SP enables users to sign in using Self Service UserID and PIN.
  The Service URL format is:
  http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#&loginType=SP.
- loginType=UID enables users to sign in using User ID and PIN.
  The Service URL format is:
  http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#&loginType=UID.

The loginType parameter can also be appended to a secure URL. If you do not append loginType to the end of the URL, the default sign in option displayed is User ID and PIN.

Step 7  Use the default values for the Service Category and Service Type fields.

Step 8  Check the Enable check box.

Step 9  (Optional) Check the Enterprise Subscription check box to subscribe all phones and device profiles to this phone service.

  Note  If you check this check box when configuring the service for the first time, you will set up this IP phone service as an enterprise subscription service. All phones and device profiles in the enterprise will automatically subscribe to this IP phone service, removing the need for you to subscribe them individually.

Step 10  Click Save.

What to do next
Configure a Device Profile for Extension Mobility Cross Cluster, on page 417

Configure a Device Profile for Extension Mobility Cross Cluster

Create a device profile to map settings onto a real device when a user logs in to Extension Mobility cross cluster.

Before you begin
Configure the Extension Mobility Phone Service, on page 416
Procedure

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

**Step 2** Perform one of the following tasks:

- To modify an existing device profile, enter search criteria, click **Find**, and click a device profile name in the resulting list.
- To add a new device profile, click **Add New**, click **Next**, choose a device profile type, click **Next**, choose a protocol, then click **Next**.

**Step 3** Configure the fields on the **Device Profile Configuration** window. See the Related Topics for more information about the fields and their configuration options.

**Step 4** Click **Save**.

**Step 5** Add a directory number (DN) to the new device profile.

---

**What to do next**

Enable Extension Mobility Cross Cluster for a User, on page 423

**Related Topics**

Device Profile Fields for Extension Mobility Cross Cluster, on page 418

---

**Device Profile Fields for Extension Mobility Cross Cluster**

**Table 32: Device Profile Settings**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product Type</td>
<td>This field displays the product type to which this device profile applies.</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>This field displays the device protocol to which this device profile applies.</td>
</tr>
<tr>
<td>Device Profile Name</td>
<td>Enter a unique name. This name can comprise up to 50 characters in length.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description of the device profile. For text, use anything that describes this particular user device profile.</td>
</tr>
</tbody>
</table>

**User Hold MOH Audio Source**

To specify the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list box.

If you do not choose an audio source, Cisco Unified Communications Manager uses the audio source that is defined in the device pool or the system default if the device pool does not specify an audio source ID.

**Note**

You define audio sources in the Music On Hold Audio Source Configuration window. For access, choose **Media Resources > Music On Hold Audio Source**.
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| User Locale        | From the drop-down list box, choose the locale that is associated with the phone user interface. The user locale identifies a set of detailed information, including language and font, to support users. Cisco Unified Communications Manager makes this field available only for phone models that support localization.  
  **Note**  
  If no user locale is specified, Cisco Unified Communications Manager uses the user locale that is associated with the device pool.  
  **Note**  
  If the users require information to display (on the phone) in any language other than English, verify that the locale installer is installed before configuring user locale. See the Cisco Unified Communications Manager Locale Installer documentation. |
| Phone Button Template | From the Phone Button Template drop-down list, choose a phone button template.  
  **Tip**  
  If you want to configure BLF/SpeedDials for the profile for presence monitoring, choose a phone button template that you configured for BLF/SpeedDials. After you save the configuration, the Add a New BLF SD link displays in the Association Information pane. For more information on BLF/SpeedDials, see the Cisco Unified Communications Manager Features and Services Guide. |
| Softkey Template   | From the Softkey Template drop-down list box, choose the softkey template from the list that displays.                                                                                                                                                                                                                   |
| Privacy            | From the Privacy drop-down list box, choose On for each phone on which you want privacy. For more information, see the Cisco Unified Communications Manager Features and Services Guide.                                                                                                                                                        |
| Single Button Barge | From the drop-down list box, choose from the following options:  
  • Off—This device does not allow users to use the Single Button Barge/cBarge feature.  
  • Barge—Choosing this option allows users to press the Single Button Barge shared-line button on the phone to barge into a call using Barge.  
  • Default—This device inherits the Single Button Barge/cBarge setting from the service parameter and device pool settings.  
  **Note**  
  If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting.  
  For more information, see the Cisco Unified Communications Manager Features and Services Guide. |
From the drop-down list box, choose from the following options:

- **Off**—This device does not allow users to use the Join Across Lines feature.
- **On**—This device allows users to join calls across multiple lines.
- **Default**—This device inherits the Join Across Lines setting from the service parameter and device pool settings.

**Note** If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting.

For more information, see the Cisco Unified Communications Manager System Guide.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Join Across Lines</td>
<td>From the drop-down list box, choose from the following options:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Off</strong>—This device does not allow users to use the Join Across Lines feature.</td>
</tr>
<tr>
<td></td>
<td>• <strong>On</strong>—This device allows users to join calls across multiple lines.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Default</strong>—This device inherits the Join Across Lines setting from the service parameter and device pool settings.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting.</td>
</tr>
</tbody>
</table>

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.

Always Use Prime Line for Voice Message

From the drop-down list box, choose one of the following options:

- **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.
- **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.
- **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.
### Field Description

**Ignore Presentation Indicators (internal calls only)**

To configure call display restrictions and ignore any presentation restriction that is received for internal calls, check the “Ignore Presentation Indicators (internal calls only)” check box.

**Tip**

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. For more information about call display restrictions, see the *Cisco Unified Communications Manager Features and Services Guide*.

**Do Not Disturb**

Check this check box to enable Do Not Disturb.

**DND Option**

When you enable DND on the phone, this parameter allows you to specify how the DND feature handles incoming calls:

- **Call Reject**—This 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.

- **Ringer Off**—This option turns off the ringer, but incoming call information gets presented to the device, so that the user can accept the call.

- **Use Common Phone Profile Setting**—This option specifies that the DND Option setting from the Common Phone Profile window will get used for this device.

**Note**

For 7940/7960 phones that are running SCCP, you can only choose the Ringer Off option. For mobile devices and dual-mode phones, you can only choose the Call Reject option. When you activate DND Call Reject on a mobile device or dual-mode phone, no call information gets presented to the device.

**DND Incoming Call Alert**

When you enable the DND Ringer Off or Call Reject option, this parameter specifies how a call displays on a phone.

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 will get 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 phone to play a beep tone only.

- **Flash Only**—For an incoming call, this option causes the phone to display a flash alert.
**Field** | **Description**  
--- | ---  
Extension Mobility Cross Cluster CSS | From the drop-down list box, choose an existing Calling Search Space (CSS) to use for this device profile for the Extension Mobility Cross Cluster feature. (To configure a new CSS or modify an existing CSS, choose Call Routing > Class of Control > Calling Search Space in Cisco Unified Communications Manager Administration.) Default value specifies None. The home administrator specifies this CSS, which gets used as the device CSS that gets assigned to the phone when the user logs in to this remote phone. For more information, see the Cisco Unified Communications Manager Features and Services Guide.  
Module 1 | You can configure one or two expansion modules for this device profile by choosing phone templates from the expansion module drop-down lists in the expansion module fields.  
**Note** You can view a phone button list at any time by choosing the View button list link next to the phone button template fields. A separate dialog box pops up and displays the phone buttons for that particular expansion module. Choose the appropriate expansion module or None.  
Module 2 | Choose the appropriate expansion module or None.  
MLPP Domain | If this user device profile will be used for MLPP precedence calls, choose the MLPP Domain from the drop-down list box.  
**Note** You define MLPP domains in the MLPP Domain Configuration window. For access, choose System > MLPP Domain.  
MLPP Indication | If this user device profile will be used for MLPP precedence calls, assign an MLPP Indication setting to the device profile. 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 box, choose a setting to assign to this device profile from the following options:  
1. Default—This device profile inherits its MLPP indication setting from the device pool of the associated device.  
2. Off—This device does not handle nor process indication of an MLPP precedence call.  
3. On—This device profile does handle and process indication of an MLPP precedence call.  
**Note** Do not configure a 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.
### Field | Description
--- | ---
MLPP Preemption | If this user device profile will be used for MLPP precedence calls, assign an MLPP Preemption setting to the device profile. 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 box, choose a setting to assign to this device profile from the following options:
1. Default—This device profile inherits its MLPP preemption setting from the device pool of the associated device.
2. Disabled—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.
3. Forceful—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.  
**Note** Do not configure a device 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.
Login User Id | From the Login User ID drop-down list box, choose a valid login user ID. 
**Note** If the device profile is used as a logout profile, specify the login user ID that will be associated with the phone. After the user logs out from this user device profile, the phone will automatically log in to this login user ID.

## Enable Extension Mobility Cross Cluster for a User

### Before you begin
Configure a Device Profile for Extension Mobility Cross Cluster, on page 417

### Procedure

**Step 1** From Cisco Unified CM Administration, choose User Management > End User.
**Step 2** Perform one of the following tasks:
- To modify the settings for an existing user, enter search criteria, click Find, and choosing an existing user from the resulting list.
- To add a new user, click Add New.
**Step 3** In the Extension Mobility pane, check the Enable Extension Mobility Cross Cluster check box.
**Step 4** Choose the device profile from the Available Profiles list pane in the Extension Mobility pane.
**Step 5** Move the device profile to the Controlled Profiles list pane.
**Step 6** Click Save.
What to do next

Subscribe Devices to Extension Mobility, on page 424

Subscribe Devices to Extension Mobility

Enable Extension Mobility on devices and subscribe to the service if you have not set up an enterprise subscription for all devices.

Before you begin

Enable Extension Mobility Cross Cluster for a User, on page 423

Procedure

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

Step 2
Find the phone on which users can use Extension Mobility Cross Cluster.

Step 3
For this device, check the Enable Extension Mobility check box in the Extension Information pane.

Step 4
In the Phone Configuration window, choose the Subscribe/Unsubscribe Services option in the Related Links drop-down list.

Step 5
Click Go.

Step 6
In the popup window that opens, choose the Extension Mobility service in the Select a Service drop-down list.

Step 7
Click Next.

Step 8
Click Subscribe.

Step 9
From the popup window, click Save, and then close the window.

Step 10
In the Phone Configuration window, click Save.

Step 11
Click OK if prompted.

Configure Certificates for Extension Mobility Cross Cluster

To configure the home and remote clusters properly, you must export certificates on each cluster to the same SFTP server and SFTP directory and consolidate them on one of the participating clusters. This procedure ensures that trust is established between the two clusters.

Before you begin

Configure Extension Mobility, on page 415

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Activate the Bulk Provisioning Service, on page 425</td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Configure Bulk Certificate Management and Export Certificates, on page 425</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Configure bulk certificate management in Cisco Unified OS Administration to export the certificates from both the home and remote clusters.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 3</th>
<th>Consolidate the Certificates, on page 426</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Consolidate certificates when all participating clusters have exported their certificates. This option is available only if two or more clusters exported their certificates to the SFTP server.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 4</th>
<th>Import the Certificates in to the Clusters, on page 427</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Import the certificates back into the home and remote (visiting) clusters.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>After an upgrade, these certificates are preserved. You do not need to reimport or reconsolidate certificates.</td>
</tr>
</tbody>
</table>

### Activate the Bulk Provisioning Service

**Before you begin**

Configure Extension Mobility, on page 415

**Procedure**

1. From Cisco Unified Serviceability, choose **Tools > Service Activation**.
2. From the **Server** drop-down list, choose the publisher node.
3. Check the **Cisco Bulk Provisioning Service** check box.
4. Click **Save**.
5. Click **OK**.

**What to do next**

Configure Bulk Certificate Management and Export Certificates, on page 425

**Configure Bulk Certificate Management and Export Certificates**

Configure bulk certificate management in Cisco Unified OS Administration to export the certificates from both the home and remote clusters.

This procedure creates a PKCS12 file that contains certificates for all nodes in the cluster.
Note

- Every participating cluster must export certificates to the same SFTP server and SFTP directory.
- You must export certificates on the cluster whenever the Tomcat, Tomcat-ECDSA, TFTP, or CAPF certificates are regenerated on any of the cluster nodes.

Before you begin

Activate the Bulk Provisioning Service, on page 425

Procedure

Step 1
From Cisco Unified OS Administration, choose Security > Bulk Certificate Management.

Step 2
Configure the settings for a TFTP server that both the home and remote clusters can reach. See the online help for information about the fields and their configuration options.

Step 3
Click Save.

Step 4
Click Export.

Step 5
In the Bulk Certificate Export window, choose All for the Certificate Type field.

Step 6
Click Export.

Step 7
Click Close.

Note
When the bulk certificate export is performed, the certificates are then uploaded to the remote cluster as follows:

- CAPF certificate gets uploaded as a CallManager-trust
- Tomcat certificate gets uploaded as a Tomcat-trust
- CallManager certificate gets uploaded as a CallManager-trust
- CallManager certificate gets uploaded as a Phone-SAST-trust

What to do next

Consolidate the Certificates, on page 426

Consolidate the Certificates

Consolidate certificates when all participating clusters have exported their certificates. This option is available only if two or more clusters exported their certificates to the SFTP server.

This procedure consolidates all PKCS12 files in the SFTP server to form a single file.

Note
If you export new certificates after consolidation, you must perform this procedure again to include the newly exported certificates.
Before you begin

Configure Bulk Certificate Management and Export Certificates, on page 425

Procedure

Step 1

Step 2
In the Certificate Type field, choose All.

Step 3
Click Consolidate.

Note
When the bulk certificate consolidate is performed, the certificates are then uploaded to the remote cluster as follows:

- CAPF certificate gets uploaded as a CallManager-trust
- Tomcat certificate gets uploaded as a Tomcat-trust
- CallManager certificate gets uploaded as a CallManager-trust
- CallManager certificate gets uploaded as a Phone-SAST-trust

What to do next

Import the Certificates into the Clusters, on page 427

Import the Certificates into the Clusters

Import the certificates back into the home and remote (visiting) clusters.

Note
After an upgrade, these certificates are preserved. You do not need to reimport or reconsolidate certificates.

Caution
After you import the certificates, the phones on the cluster will automatically restart.

Before you begin

Consolidate the Certificates, on page 426

Procedure

Step 1

Step 2
From the Certificate Type drop-down list, choose All.

Step 3
Choose Import.
When the bulk certificate import is performed, the certificates are then uploaded to the remote cluster as follows:

- CAPF certificate gets uploaded as a CallManager-trust
- Tomcat certificate gets uploaded as a Tomcat-trust
- CallManager certificate gets uploaded as a CallManager-trust
- CallManager certificate gets uploaded as a Phone-SAST-trust

The following types of certificates determines phones that are restarted:

- Callmanager - ALL phones only IF TFTP service is activated on the node the certificate belongs.
- TVS - SOME phones based on Callmanager group membership.
- CAPF - ALL phones only IF CAPF is activated.

---

**Configure Extension Mobility Cross Cluster Devices and Templates**

**Before you begin**

Configure Certificates for Extension Mobility Cross Cluster, on page 424

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Create a Common Device Configuration, on page 429</td>
<td>Configure a common device configuration to specify the services or features that will be associated with a particular user.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure an Extension Mobility Cross Cluster Template, on page 429</td>
<td>Create an extension mobility cross cluster template to link the common device configuration with this feature.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Set the Default Template, on page 430</td>
<td>Set the extension mobility cross cluster template that you created as the default template.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Add Extension Mobility Cross Cluster Devices, on page 430</td>
<td>Insert extension mobility cross cluster device entries into your system database. Each device is identified with a unique name in the format EMCC1, EMCC2, and so on. The Bulk Administration Tool assigns device numbers by obtaining the last one used.</td>
</tr>
</tbody>
</table>
Create a Common Device Configuration

Configure a common device configuration to specify the services or features that will be associated with a particular user.

**Before you begin**

Configure Certificates for Extension Mobility Cross Cluster, on page 424

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Device Settings &gt; Common Device Configuration</strong>.</th>
</tr>
</thead>
</table>
| Step 2 | Perform one of the following tasks:  
  - To modify an existing common device configuration, enter search criteria, click **Find**, and choose a common device configuration from the resulting list.  
  - To add a new common device configuration, click **Add New**. |
| Step 3 | Configure the fields on the **Common Device Configuration** window. See the online help for more information about the fields and their configuration options. |
| Step 4 | Click **Save**. |

**What to do next**

Configure an Extension Mobility Cross Cluster Template, on page 429

Configure an Extension Mobility Cross Cluster Template

Create an extension mobility cross cluster template to link the common device configuration with this feature.

**Before you begin**

Create a Common Device Configuration, on page 429

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Bulk Administration &gt; EMCC &gt; EMCC Template</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Add New</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the fields on the <strong>EMCC Template Configuration</strong> window. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**What to do next**

Set the Default Template, on page 430
Set the Default Template

Set the extension mobility cross cluster template that you created as the default template.

**Before you begin**
Configure an Extension Mobility Cross Cluster Template, on page 429

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Bulk Administration &gt; EMCC &gt; Insert/Update EMCC.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Update EMCC Devices.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Default EMCC Template drop-down list, choose the extension mobility cross cluster device</td>
</tr>
<tr>
<td></td>
<td>template that you configured.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Run Immediately.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Submit.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Verify the success of the job:</td>
</tr>
<tr>
<td></td>
<td>a) Choose Bulk Administration &gt; Job Scheduler.</td>
</tr>
<tr>
<td></td>
<td>b) Locate the Job ID of your job.</td>
</tr>
</tbody>
</table>

**What to do next**
Add Extension Mobility Cross Cluster Devices, on page 430

Add Extension Mobility Cross Cluster Devices

Insert extension mobility cross cluster device entries into your system database. Each device is identified with a unique name in the format EMCC1, EMCC2, and so on. The Bulk Administration Tool assigns device numbers by obtaining the last one used.

**Before you begin**
Set the Default Template, on page 430

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Bulk Administration &gt; EMCC &gt; Insert/Update EMCC.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Insert EMCC Devices.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Enter the number of devices you are adding in the Number of EMCC Devices to be added field.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Run Immediately and click Submit.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Refresh the window and verify that the Number of EMCC Devices already in database value shows</td>
</tr>
<tr>
<td></td>
<td>the number of devices that you added.</td>
</tr>
</tbody>
</table>
Configure a Geolocation Filter for Extension Mobility Cross Cluster

Configure a geolocation filter to specify criteria for device location matching, such as country, state, and city values. Geolocations are used to identify the location of a device, and the filter indicates what parts of the geolocation are significant.

Before you begin
Configure Extension Mobility Cross Cluster Devices and Templates, on page 428

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose System &gt; Geolocation Filter.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the fields on the Geolocation Filter Configuration window. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

What to do next
Configure Feature Parameters for Extension Mobility Cross Cluster, on page 431

Configure Feature Parameters for Extension Mobility Cross Cluster

Select values for the feature parameters that you configured, such as the geolocation filter.

Before you begin
Configure Bulk Certificate Management and Export Certificates, on page 425

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Advanced Features &gt; EMCC &gt; EMCC Feature Configuration.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure the fields on the EMCC Feature Configuration window. See the Related Topics section for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

What to do next
Configure Intercluster SIP Trunk for Extension Mobility Cross Cluster, on page 435

Related Topics
Feature Parameter Fields for Extension Mobility Cross Cluster, on page 432
### Feature Parameter Fields for Extension Mobility Cross Cluster

**Table 33: Feature Parameter Fields for Extension Mobility Cross Cluster**

<table>
<thead>
<tr>
<th>EMCC Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default TFTP Server for EMCC Login Device</td>
<td>Choose the computer name or IP address of the default TFTP server that devices logging into extension mobility cross cluster (EMCC) from a remote cluster should use.</td>
</tr>
<tr>
<td>Backup TFTP Server for EMCC Login Device</td>
<td>Choose the computer name or IP address of the backup TFTP server that devices logging into EMCC from a remote cluster should use.</td>
</tr>
<tr>
<td>Default Interval for Expired EMCC Device Maintenance</td>
<td>Specify the number of minutes that elapse between system checks for expired EMCC devices. An expired EMCC device is a device that logged in to EMCC from a remote cluster, but that, because of a WAN failure or a connectivity issue, the phone logged out of the visiting cluster. When connectivity was restored, the device logged back into the visiting cluster. During this maintenance job, the Cisco Extension Mobility service checks the Cisco Unified Communications Manager database for any expired EMCC devices and automatically logs them out. The default value is 1440 minutes. Valid values range from 10 minutes to 1440 minutes.</td>
</tr>
<tr>
<td>Enable All Remote Cluster Services When Adding A New Remote Cluster</td>
<td>Choose whether you want all services on a new remote cluster to be automatically enabled when you add a new cluster. Valid values are True (enable all services on the remote cluster automatically) or False (manually enable the services on the remote cluster via the Remote Cluster Configuration window in Cisco Unified Communications Manager Administration). You can enable the services manually so that you have time to configure the EMCC feature completely before enabling the remote services. The default value is False.</td>
</tr>
</tbody>
</table>
| CSS for PSTN Access SIP Trunk                        | Choose the calling search space (CSS) that the PSTN Access SIP trunk for processing EMCC calls uses. The PSTN Access SIP trunk is the SIP trunk that you configured for PSTN access in the Intercluster Service Profile window. Calls over this trunk are intended for and are routed to only the local PSTN that is co-located with the EMCC logged-in phone that initiates the call. Valid values are the following:  
  - Use Trunk CSS (PSTN calls use the local route group, which can prove useful for properly routing emergency service calls)  
  - Use phone's original device CSS (PSTN calls are routed using the configured calling search space on the remote phone, that is, the CSS that is used when the phone is not logged into EMCC).  
  The default value is Use trunk CSS. |
**EMCC Parameter** | **Description**
--- | ---
EMCC Geolocation Filter | Choose the geolocation filter that you have configured for use EMCC. Based on the information in the geolocation that associates with a phone that is logged in through Extension Mobility from another cluster, as well as the selected EMCC geolocation filter, Cisco Unified Communications Manager places the phone into a roaming device pool. Cisco Unified Communications Manager determines which roaming device pool to use by evaluating which device pool best matches the phone geolocation information after the EMCC geolocation filter is applied.

| EMCC Region Max Audio Bit Rate | This parameter specifies the maximum audio bit rate for all EMCC calls, regardless of the region associated with the other party. The default value is 8 kbps (G.729). **Note** All participating EMCC clusters must specify the same value for the EMCC region max audio bit rate.

<p>| EMCC Region Max Video Call Bit Rate (Includes Audio) | This parameter specifies the maximum video call bit rate for all EMCC video calls, regardless of the maximum video call bit rate of the region associated with the other party. The default value is 384. Valid values range from 0 to 8128. <strong>Note</strong> All participating EMCC clusters must specify the same value for the EMCC region max video call bit rate. |</p>
<table>
<thead>
<tr>
<th><strong>EMCC Parameter</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>EMCC Region Link Loss Type</td>
<td>This parameter specifies the link loss type between any EMCC phone and devices in any remote cluster.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>To allow two-way audio on EMCC calls, all participating EMCC clusters must use the same EMCC region link loss type.</td>
</tr>
<tr>
<td></td>
<td>Based on the option that you choose, Cisco Unified Communications Manager attempts to use the optimal audio codec for the EMCC call while observing the configured EMCC region max audio bit rate.</td>
</tr>
<tr>
<td></td>
<td>Valid values are the following:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Lossy</strong>—A link where some packet loss can or may occur, for example, DSL.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Low Loss</strong>—A link where low packet loss occurs, for example, T1.</td>
</tr>
<tr>
<td></td>
<td>When you set this parameter to <strong>Lossy</strong>, Cisco Unified Communications Manager chooses the optimal codec within the limit that is set by the EMCC Region Max Audio Bit Rate, based on audio quality. Some packet loss will occur.</td>
</tr>
<tr>
<td></td>
<td>When this parameter is set to <strong>Low Loss</strong>, Cisco Unified Communications Manager chooses the optimal codec within the limit that is set by the EMCC Region Max Audio Bit Rate, based on audio quality. Little or no packet loss will occur.</td>
</tr>
<tr>
<td></td>
<td>The only difference in the audio codec preference ordering between the low loss and lossy options is that G.722 is preferred over Internet Speech Audio Codec (iSAC) when the link loss type is set as low loss, whereas iSAC is preferred over G.722 when the link loss type is set as lossy.</td>
</tr>
<tr>
<td></td>
<td>The default value is <strong>Low Loss</strong>.</td>
</tr>
<tr>
<td>RSVP SIP Trunk KeepAlive Timer</td>
<td>Specify the number of seconds that Cisco Unified Communications Manager waits between sending or receiving KeepAlive messages or acknowledgments between two clusters over EMCC RSVP SIP trunks.</td>
</tr>
<tr>
<td></td>
<td>An EMCC RSVP SIP trunk is a SIP trunk that has Cisco Extension Mobility Cross Cluster configured as the Trunk Service Type and that has been selected as the SIP Trunk for RSVP Agent in the Intercluster Service Profile window.</td>
</tr>
<tr>
<td></td>
<td>When two of these intervals elapse without receipt of a KeepAlive message or an acknowledgment, Cisco Unified Communications Manager releases the RSVP resources with the remote cluster.</td>
</tr>
<tr>
<td></td>
<td>The default value is 15 seconds. Valid values range from 1 second to 600 seconds.</td>
</tr>
<tr>
<td>Default Server For Remote Cluster Update</td>
<td>Choose the default server name or IP address of the primary node in this local cluster that has the Cisco Extension Mobility service activated. The remote cluster accesses this node to get information about this local cluster.</td>
</tr>
<tr>
<td>Backup Server for Remote Cluster Update</td>
<td>Choose the default server name or IP address of the secondary node in this local cluster that has the Cisco Extension Mobility service activated. The remote cluster accesses this node when the primary node is down to retrieve information about this local cluster.</td>
</tr>
</tbody>
</table>
**EMCC Parameter** | **Description**
--- | ---
Remote Cluster Update Interval | Specify an interval, in minutes, during which the Cisco Extension Mobility service on the local node collects information about the remote EMCC cluster. Collected information includes such details as the remote cluster Cisco Unified Communications Manager version and service information. The default value is 30. Valid values range from 15 minutes to 10,080 minutes.

### Configure Intercluster SIP Trunk for Extension Mobility Cross Cluster

Configure trunks to process inbound or outbound traffic for intercluster PSTN access and RSVP agent services. You can configure one trunk for both PSTN access and RSVP agent services or one trunk for each service. You do not need more than two SIP trunks for extension mobility cross cluster.

**Before you begin**
Configure Feature Parameters for Extension Mobility Cross Cluster, on page 431

**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 **Trunk Service Type** drop-down list, choose **Extension Mobility Cross Clusters**.
**Step 5** Click **Next**.
**Step 6** Configure the fields in the **Trunk Configuration** window. See the online help for more information about the fields and their configuration options.
**Step 7** Click **Save**.

**What to do next**
Configure an Intercluster Service Profile for Extension Mobility Cross Cluster, on page 435

### Configure an Intercluster Service Profile for Extension Mobility Cross Cluster

Configure the intercluster service profile to activate extension mobility cross cluster. The profile collects all the configuration that precedes and provides a results report.

**Before you begin**
Configure Intercluster SIP Trunk for Extension Mobility Cross Cluster, on page 435
Configure Remote Cluster Services

Configure the remote cluster for extension mobility cross cluster. This step completes the link between the home cluster with remote (visiting) cluster.

Before you begin

Configure an Intercluster Service Profile for Extension Mobility Cross Cluster, on page 435

Procedure

Step 1 From Cisco Unified CM Administration, choose Advanced Features > EMCC > EMCC Intercluster Service Profile.
Step 2 Configure the fields on the EMCC Intercluster Service Profile Configuration window. See the online help for more information about the fields and their configuration options.
Step 3 If no failure messages appear in the popup window, click Save.

What to do next

Configure Remote Cluster Services, on page 436

Extension Mobility Cross Cluster Interactions and Restrictions

- Extension Mobility Cross Cluster Restrictions, on page 437
### Extension Mobility Cross Cluster Interactions

#### Table 34: Extension Mobility Cross Cluster Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio</td>
<td>The default maximum audio bit-rate for EMCC login device is set to 8 kbps (G.729).</td>
</tr>
<tr>
<td>Call Admission Control (CAC)</td>
<td>• The home cluster is unaware of visiting cluster locations and regions. &lt;br&gt;• The system cannot apply Cisco Unified Communications Manager locations and regions across the cluster boundaries. &lt;br&gt;• RSVP agent-based CAC uses RSVP agents in the visiting cluster.</td>
</tr>
<tr>
<td>Call Forwarding</td>
<td>EMCC supports call forwarding.</td>
</tr>
<tr>
<td>Cisco Extension Mobility login and logout</td>
<td>User authentication takes place across clusters.</td>
</tr>
<tr>
<td>Media resources for the visiting phone</td>
<td>Examples include RSVP agent, TRP, music on mold (MOH), MTP, transcoder, and conference bridge. Media resources are local to the visiting phone (other than RSVP agents).</td>
</tr>
<tr>
<td>PSTN access for the visiting phone</td>
<td>• E911 calls are routed to the local gateways of the PSTN. &lt;br&gt;• Local calls are routed to the local gateways of the PSTN. &lt;br&gt;• Calls terminating to local route groups route to local gateways in the visiting cluster.</td>
</tr>
<tr>
<td>Other call features and services</td>
<td>Example restriction: Intercom configuration specifies configuration to a static device, so EMCC does not support the intercom feature.</td>
</tr>
<tr>
<td>Security</td>
<td>• Cross-cluster security is provided by default. &lt;br&gt;• Cisco Unified IP Phones with secure and nonsecure phone security profiles are supported.</td>
</tr>
</tbody>
</table>

### Extension Mobility Cross Cluster Restrictions

#### Table 35: Extension Mobility Cross Cluster Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unsupported Features</td>
<td>• EMCC does not support the intercom feature, because intercom configuration requires a static device. &lt;br&gt;• Location CAC is not supported, but RSVP-based CAC is supported.</td>
</tr>
</tbody>
</table>
## Extension Mobility Cross Cluster Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>EMCC Device Cannot Be Provisioned in More Than One Cluster</td>
<td>For EMCC to function properly, you cannot configure the same phone (device name) in two clusters. Otherwise, login will fail due to the duplicate device error (37). For this reason, for cluster deployed with EMCC you should disable Autoregistration on all Unified Communication Manager nodes to prevent a new device being created in the home cluster after EMCC logout.</td>
</tr>
<tr>
<td>Number of EMCC Devices</td>
<td>Cisco Unified Communications Manager can support a MaxPhones value of 60,000. Include EMCC in the total number of devices that are supported in the cluster by using the following calculation: Phones + (2 x EMCC devices) = MaxPhones</td>
</tr>
<tr>
<td>Note</td>
<td>EMCC login does not affect the number of licenses that are used in the home cluster.</td>
</tr>
<tr>
<td>Visiting Device Logout Limitations</td>
<td>• If the home cluster administrator disables EMCC for a user while the user is logged in with EMCC, the system does not automatically log this user out. Instead, the system does not allow future EMCC attempts by this user. The current EMCC session continues until the user logs out.</td>
</tr>
<tr>
<td></td>
<td>• In the visiting cluster, the <strong>Phone Configuration</strong> window has a Log Out button for extension mobility. This button is also used by the visiting cluster administrator to log out an EMCC phone. Because the EMCC phone is not currently registered with the visiting Cisco Unified Communications Manager, this operation is like a database cleanup in the visiting cluster. The EMCC phone remains registered with the home Cisco Unified Communications Manager until the phone returns to the visiting cluster because of a reset or a logout from the home cluster.</td>
</tr>
</tbody>
</table>
The extension mobility service in participating clusters performs a periodic remote cluster update. The **Remote Cluster Update Interval** feature parameter controls the update interval. The default interval is 30 minutes.

If the extension mobility service on cluster A does not receive a reply from a remote cluster (such as cluster B) for this update, the Remote Cluster window for cluster A shows that the Remote Activated service is set to False for cluster B.

In this case, the visiting cluster does not receive any response from the home cluster and sets the Remote Activated values for the home cluster to False.

During this interval, a visiting phone may not be able to log in by using EMCC. The visiting phone receives the “Login is unavailable” error message.

At this point, a login attempt to EMCC from a visiting phone can fail; the phone displays the “Login is unavailable” error message. This error occurs because the visiting cluster has not yet detected the change of the home cluster from out-of-service to in-service.

Remote cluster status change is based on the value of the Remote Cluster Update Interval EMCC feature parameter and on when the visiting extension mobility service performed the last query or update.

You can select **Update Remote Cluster Now** in the **Remote Cluster Service Configuration** window (Advanced Features > EMCC > EMCC Remote Cluster) to change Remote Activate values to True, which also allows EMCC logins. Otherwise, after the next periodic update cycle, EMCC logins by visiting phones will return to normal.

### EMCC Login Result for Different Cluster Versions with loginType

The following table shows the login result of the Extension Mobility Cross Cluster feature for different cluster versions when the `loginType` parameter is used in the service URL.

<table>
<thead>
<tr>
<th>Visiting Cluster Version</th>
<th>Home Cluster Version</th>
<th>loginType in Visiting Cluster EM URL*</th>
<th>EMCC Login Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.0</td>
<td>12.0</td>
<td>Not mentioned (Default URL)</td>
<td>Success</td>
</tr>
<tr>
<td>12.0</td>
<td>12.0</td>
<td>UID, SP, or DN</td>
<td>Success</td>
</tr>
<tr>
<td>12.0</td>
<td>11.5 and below</td>
<td>Not mentioned (Default URL)</td>
<td>Success</td>
</tr>
<tr>
<td>12.0</td>
<td>11.5 and below</td>
<td>UID, SP, or DN</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Fail with error code - 1</td>
<td></td>
</tr>
<tr>
<td>11.5 and below</td>
<td>12.0</td>
<td>Not mentioned (Default URL)</td>
<td>Success</td>
</tr>
<tr>
<td>Visiting Cluster Version</td>
<td>Home Cluster Version</td>
<td>loginType in Visiting Cluster EM URL*</td>
<td>EMCC Login Result</td>
</tr>
<tr>
<td>-------------------------</td>
<td>---------------------</td>
<td>--------------------------------------</td>
<td>-------------------</td>
</tr>
<tr>
<td>11.5 and below</td>
<td>12.0</td>
<td>UID, SP, or DN ***</td>
<td>Success</td>
</tr>
</tbody>
</table>

* Following are the loginType parameter options:
  - UID—Users login using User ID and PIN
  - SP—Users login using Self Service User ID and PIN
  - DN—Users login using Primary Extension and PIN

** Fail with error code - 1 — (When EMService could not parse the XML request from EMApp/EMService)

*** loginType will be ignored and UserID / PIN login prompt gets populated on the phone

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**Extension Mobility Cross Cluster and Security Mode for Different Cluster Versions**

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Note

Phone configuration files can be encrypted only if both the home cluster and visiting cluster versions are 9.x or later, and when the TFTP encryption configuration flag is enabled.

During EMCC login, if both the visiting cluster and home cluster versions are in 9.x or later, the phone will behave in various modes as shown in the following table.

**Table 37: Supported Security Modes When Both Visiting Cluster and Home Cluster Are In 9.x or later Versions**

<table>
<thead>
<tr>
<th>Home Cluster Version</th>
<th>Home Cluster Mode</th>
<th>Visiting Cluster Version</th>
<th>Visiting Cluster Mode</th>
<th>Visiting Phone Mode</th>
<th>EMCC Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.x or later</td>
<td>Mixed</td>
<td>9.x or later</td>
<td>Mixed</td>
<td>Secure</td>
<td>Secure EMCC</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Mixed</td>
<td>9.x or later</td>
<td>Mixed</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Mixed</td>
<td>9.x or later</td>
<td>Non-secure</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Non-secure</td>
<td>9.x or later</td>
<td>Mixed</td>
<td>Secure</td>
<td>Login fails</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Non-secure</td>
<td>9.x or later</td>
<td>Non-secure</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
</tbody>
</table>
During EMCC login, if the visiting cluster version is 8.x and the home cluster version is 9.x or later, the phone will behave in various modes as shown in the following table.

Table 38: Supported Security Modes When Visiting Cluster Is In 8.x and Home Cluster Is In 9.x or later Version

<table>
<thead>
<tr>
<th>Home Cluster Version</th>
<th>Home Cluster Mode</th>
<th>Visiting Cluster Version</th>
<th>Visiting Cluster Mode</th>
<th>Visiting Phone Mode</th>
<th>EMCC Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>9.x or later</td>
<td>Mixed</td>
<td>8.x</td>
<td>Mixed</td>
<td>Secure</td>
<td>Not supported</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Mixed</td>
<td>8.x</td>
<td>Mixed</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Mixed</td>
<td>8.x</td>
<td>Non-secure</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Non-secure</td>
<td>8.x</td>
<td>Mixed</td>
<td>Secure</td>
<td>Not supported</td>
</tr>
<tr>
<td>9.x or later</td>
<td>Non-secure</td>
<td>8.x</td>
<td>Non-secure</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
</tbody>
</table>

During EMCC login, if the visiting cluster version is 9.x or later and the home cluster version is 8.x, the phone will behave in various modes as shown in the following table.

Table 39: Supported Security Modes When Visiting Cluster Is In 9.x or later and Home Cluster Is In 8.x Version

<table>
<thead>
<tr>
<th>Home Cluster Version</th>
<th>Home Cluster Mode</th>
<th>Visiting Cluster Version</th>
<th>Visiting Cluster Mode</th>
<th>Visiting Phone Mode</th>
<th>EMCC Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>8.x</td>
<td>Mixed</td>
<td>9.x or later</td>
<td>Mixed</td>
<td>Secure</td>
<td>Login fails</td>
</tr>
<tr>
<td>8.x</td>
<td>Mixed</td>
<td>9.x or later</td>
<td>Mixed</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
<tr>
<td>8.x</td>
<td>Mixed</td>
<td>9.x or later</td>
<td>Non-secure</td>
<td>Non-secure</td>
<td>Non-secure EMCC</td>
</tr>
<tr>
<td>8.x</td>
<td>Non-secure</td>
<td>9.x or later</td>
<td>Mixed</td>
<td>Secure</td>
<td>Login fails</td>
</tr>
<tr>
<td>8.x</td>
<td>Non-secure</td>
<td>9.x or later</td>
<td>Non-secure</td>
<td>Secure</td>
<td>Non-secure EMCC</td>
</tr>
</tbody>
</table>
# Extension Mobility Cross Cluster Troubleshooting

## Extension Mobility Application Error Codes

<table>
<thead>
<tr>
<th>Error Code</th>
<th>Phone Display</th>
<th>Quick Description</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>201</td>
<td>Please try to login again (201)</td>
<td>Authentication Error</td>
<td>If the user is an EMCC user, this error can occur if “EMCC” is not activated on the Intercluster Service Profile window.</td>
</tr>
<tr>
<td>202</td>
<td>Please try to login again (202)</td>
<td>Blank userid or pin</td>
<td>The user enters a blank user ID or PIN.</td>
</tr>
<tr>
<td>204</td>
<td>Login is unavailable (204)</td>
<td>Directory server error</td>
<td>The EMAApp sends this error to the phone when IMS could not authenticate the user with the given PIN.</td>
</tr>
<tr>
<td>205</td>
<td>Login is unavailable (205) Logout is unavailable (205)</td>
<td>User Profile Absent</td>
<td>Occurs when the user profile information cannot be retrieved from the cache or the database.</td>
</tr>
<tr>
<td>207</td>
<td>Login is unavailable(207) Logout is unavailable(207)</td>
<td>Device Name Empty</td>
<td>Occurs when the device or name tag is missing in the request URI. This cannot happen with real devices and can occur only if the request is sent from third-party applications.</td>
</tr>
<tr>
<td>208</td>
<td>Login is unavailable(208) Logout is unavailable(208)</td>
<td>EMService Connection Error</td>
<td>The visiting EMAApp cannot connect to any Visiting EMService. (The service is down or not activated.) The visiting EMService cannot connect to the Home EMServices (the WAN is down or certificates are not trusted.)</td>
</tr>
<tr>
<td>Error Code</td>
<td>Phone Display</td>
<td>Quick Description</td>
<td>Reason</td>
</tr>
<tr>
<td>------------</td>
<td>---------------</td>
<td>-------------------</td>
<td>--------</td>
</tr>
<tr>
<td>210</td>
<td>Login is unavailable(210)</td>
<td>Init Fail-Contact Admin</td>
<td>An error (such as a database connection failure) occurred while initializing EMAp. The error can occur because of a failure to connect to the database during startup.</td>
</tr>
<tr>
<td>211</td>
<td>Login is unavailable(211)</td>
<td>EMCC Not Activated</td>
<td>Occurs when the PSTN is not activated in the Intercluster Service Profile window of the visiting cluster.</td>
</tr>
<tr>
<td>212</td>
<td>Login is unavailable(212)</td>
<td>Cluster ID is invalid</td>
<td>Occurs when a remote cluster update fails by sending an incorrect cluster ID to the remote cluster.</td>
</tr>
<tr>
<td>213</td>
<td>Login is unavailable(213)</td>
<td>Device does not support EMCC</td>
<td>Occurs when a device does not support EMCC.</td>
</tr>
</tbody>
</table>
| 215        | LoginType invalid(215) | Login Type is invalid | Occurs when `loginType` is invalid. The allowed values are:  
  - `SP` for Self-service UserID  
  - `DN` for Primary Extension  
  - `UID` for User ID |
| 216        | DN has multiple users(216) | DN has multiple users | Occurs when the Extension used for EM login is assigned for multiple users as Primary. |

### Extension Mobility Service Error Codes

*Table 41: Extension Mobility Service Error Codes*

<table>
<thead>
<tr>
<th>Error Code</th>
<th>Phone Display</th>
<th>Quick Description</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Login is unavailable(0)</td>
<td>Unknown Error</td>
<td>The EMService failed for an unknown reason.</td>
</tr>
<tr>
<td>Error Code</td>
<td>Phone Display</td>
<td>Quick Description</td>
<td>Reason</td>
</tr>
<tr>
<td>------------</td>
<td>---------------</td>
<td>-------------------</td>
<td>--------</td>
</tr>
<tr>
<td>1</td>
<td>Login is unavailable(1) Logout is unavailable(1)</td>
<td>Error on parsing</td>
<td>When the EMService cannot parse the XML request from the EMApp or EMService. This error occurs when third-party applications send an incorrect query to login XML (EM API). The error can also occur because of a version mismatch between home and visiting clusters.</td>
</tr>
<tr>
<td>2</td>
<td>Login is unavailable(2)</td>
<td>EMCC Authentication Error</td>
<td>The EMCC user credentials cannot be authenticated because the user entered an incorrect PIN.</td>
</tr>
<tr>
<td>3</td>
<td>Login is unavailable(3) Logout is unavailable(3)</td>
<td>Invalid App User</td>
<td>Invalid application user. This error commonly occurs because of the EM API.</td>
</tr>
<tr>
<td>4</td>
<td>Login is unavailable(4) Logout is unavailable(4)</td>
<td>Policy Validation error</td>
<td>The EM Service sends this error when it cannot validate the login or logout request because of an unknown reason, an error while querying the database or an error while retrieving information from the cache.</td>
</tr>
<tr>
<td>5</td>
<td>Login is unavailable(5) Logout is unavailable(5)</td>
<td>Dev. logon disabled</td>
<td>A user logs into a device that has <strong>Enable Extension Mobility</strong> unchecked in the <strong>Phone Configuration</strong> window.</td>
</tr>
<tr>
<td>6</td>
<td>Login is unavailable(6) Logout is unavailable(6)</td>
<td>Database Error</td>
<td>Whenever the database returns an exception while executing the query or stored procedure that the EM Service requests (login/logout or device/user query), the EM Service sends this error code to EMApp.</td>
</tr>
<tr>
<td>Error Code</td>
<td>Phone Display</td>
<td>Quick Description</td>
<td>Reason</td>
</tr>
<tr>
<td>------------</td>
<td>---------------</td>
<td>-------------------</td>
<td>--------</td>
</tr>
<tr>
<td>8</td>
<td>Login is unavailable(8) Logout is unavailable(8)</td>
<td>Query type undetermined</td>
<td>No valid query was sent to the EMService (DeviceUserQuery and UserDeviceQuery are valid ones). This error occurs because of the EM API or incorrect XML input.</td>
</tr>
<tr>
<td>9</td>
<td>Login is unavailable(9) Logout is unavailable(9)</td>
<td>Dir. User Info Error</td>
<td>This error appears in two cases: 1. IMS returns an exception when it attempts to authenticate a user. 2. When information about a user cannot be retrieved either from the cache or database.</td>
</tr>
<tr>
<td>10</td>
<td>Login is unavailable(10) Logout is unavailable(10)</td>
<td>User lacks app proxy rights</td>
<td>The user tries to log in on behalf of another user. By default, a CCMSysUser has administrative rights.</td>
</tr>
<tr>
<td>11</td>
<td>Login is unavailable(11) Logout is unavailable(11)</td>
<td>Device Does not exist</td>
<td>The phone record entry is absent in the device table.</td>
</tr>
<tr>
<td>12</td>
<td>Phone record entry is absent in the device table</td>
<td>Dev. Profile not found</td>
<td>No device profile is associated with the remote user.</td>
</tr>
<tr>
<td>18</td>
<td>Login is unavailable(18)</td>
<td>Another user logged in</td>
<td>Another user is already logged in on the phone.</td>
</tr>
<tr>
<td>19</td>
<td>Logout is unavailable(19)</td>
<td>No user logged in</td>
<td>The system attempted to log out a user who has not logged in. This error occurs when sending logout requests from third-party applications (EM API).</td>
</tr>
<tr>
<td>20</td>
<td>Login is unavailable(20) Logout is unavailable(20)</td>
<td>Hoteling flag error</td>
<td>Enable Extension Mobility is unchecked in the Phone Configuration window.</td>
</tr>
<tr>
<td>Error Code</td>
<td>Phone Display</td>
<td>Quick Description</td>
<td>Reason</td>
</tr>
<tr>
<td>------------</td>
<td>--------------------------------</td>
<td>----------------------------------</td>
<td>------------------------------------------------------------------------</td>
</tr>
<tr>
<td>21</td>
<td>Login is unavailable(21)</td>
<td>Hoteling Status error</td>
<td>The current user status was not retrieved from either the local cache or database.</td>
</tr>
<tr>
<td></td>
<td>Logout is unavailable(21)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>22</td>
<td>Login is unavailable(22)</td>
<td>Dev. logon disabled</td>
<td>Occurs when EM is not enabled on device and the request is sent via EM API or when the services button is pressed on phone.</td>
</tr>
<tr>
<td>23</td>
<td>Login is Unavailable (23)</td>
<td>User does not exist</td>
<td>Occurs when the given user ID is not found (in any of the remote clusters).</td>
</tr>
<tr>
<td></td>
<td>Logout is Unavailable (23)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>25</td>
<td>Multi-Login Not Allowed (25)</td>
<td>User logged in elsewhere</td>
<td>The user is currently logged in to some other phone either in the local cluster or remote cluster.</td>
</tr>
<tr>
<td>26</td>
<td>Login is unavailable(26)</td>
<td>Busy, please try again</td>
<td>Occurs when the EMService has currently reached the threshold level of Maximum Concurrent Requests service parameter.</td>
</tr>
<tr>
<td></td>
<td>Logout is unavailable(26)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>28</td>
<td>Login is unavailable(28)</td>
<td>Untrusted IP Error</td>
<td>Occurs when the Validate IPAddress service parameter is set to True and the user tries to log in or log out from a machine whose IP address is not trusted. For example, a third-party application or EM API from a machine is not listed in the Trusted List of Ips service parameter.</td>
</tr>
<tr>
<td></td>
<td>Logout is unavailable(28)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>29</td>
<td>Login is unavailable(29)</td>
<td>ris down-contact admin</td>
<td>The Real-Time Information Server Data Collector (RISDC) cache is not created or initialized, and the EMService is unable to connect to RISDC.</td>
</tr>
<tr>
<td></td>
<td>Logout is unavailable(29)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Error Code</td>
<td>Phone Display</td>
<td>Quick Description</td>
<td>Reason</td>
</tr>
<tr>
<td>------------</td>
<td>-------------------------------</td>
<td>----------------------------------------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>30</td>
<td>Login is unavailable(30)</td>
<td>Proxy not allowed</td>
<td>When login and logout occur through proxy (“Via” is set in HTTP header) and the Allow Proxy service parameter is set to False.</td>
</tr>
<tr>
<td>31</td>
<td>Login is unavailable(31)</td>
<td>EMCC Not Activated for the user</td>
<td>Occurs when the <a href="#">Enable Extension Mobility Cross Cluster</a> check box is not checked in the <a href="#">End User Configuration</a> window of the home cluster.</td>
</tr>
<tr>
<td>32</td>
<td>Login is unavailable(32)</td>
<td>Device does not support EMCC</td>
<td>Occurs when a device model does not have EMCC capability.</td>
</tr>
<tr>
<td>33</td>
<td>Login is unavailable(33)</td>
<td>No free EMCC dummy device</td>
<td>Occurs when all the EMCC dummy devices are in use by other EMCC logins.</td>
</tr>
<tr>
<td>35</td>
<td>Login is unavailable(35)</td>
<td>Visiting Cluster Information is not present in Home Cluster</td>
<td>Occurs when the home cluster does not have an entry for this visiting cluster.</td>
</tr>
<tr>
<td>36</td>
<td>Login is unavailable(36)</td>
<td>No Remote Cluster</td>
<td>Occurs when the administrator has not added a remote cluster.</td>
</tr>
<tr>
<td>37</td>
<td>Login is Unavailable (37)</td>
<td>Duplicate Device Name</td>
<td>Occurs when the same device name exists in both the home cluster and visiting cluster.</td>
</tr>
<tr>
<td>38</td>
<td>Login is unavailable(38)</td>
<td>EMCC Not Allowed</td>
<td>Occurs when the home cluster does not want to allow EMCC login (The <a href="#">Enable Extension Mobility Cross Cluster</a> check box is not checked in the home cluster).</td>
</tr>
</tbody>
</table>

---

**Feature Configuration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1**

**Extension Mobility Service Error Codes**

---

447
<table>
<thead>
<tr>
<th>Error Code</th>
<th>Phone Display</th>
<th>Quick Description</th>
<th>Reason</th>
</tr>
</thead>
</table>
| 39         | Please try to login again (201) | Configuration Issue | Occurs when the Default TFTP Server and Backup TFTP Server for EMCC login device are not set properly in EMCC Feature Configuration Page.  
**Note** This is internal error code. |
| 40         | Please try to login again (23) | No Response from Remote Host | Occurs when response not getting from Remote Host.  
**Note** This is internal error code. |
| 41         | PIN change is required | PIN change is required | Occurs when admin enables User Must Change at Next Login for PIN. In that case user is redirected to Change credentials page.  
**Note** This is internal error code. |
| 42         | Login is unavailable(42)  
Logout is unavailable(42) | Invalid ClusterID | Occurs when the remote cluster ID is not valid.  
This error can occur during a remote cluster update. |
| 43         | Login is unavailable(43) | Device Security mode error | The Device Security Profile that is associated to the EMCC device should be set to Nonsecure for its Device Security Mode. |
| 44         | Please try to login again (201) | Configuration Issue | Occurs when the cluster ID is not valid.  
**Note** This is internal error code. |
<table>
<thead>
<tr>
<th>Error Code</th>
<th>Phone Display</th>
<th>Quick Description</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>45</td>
<td>Login is unsuccessful(45)</td>
<td>Remote Cluster version not supported</td>
<td>Occurs during EMCC login when the visiting cluster version is 9.x and is in mixed mode, the phone is in secure mode, and the home cluster version is 8.x.</td>
</tr>
<tr>
<td>46</td>
<td>Login is unsuccessful(46)</td>
<td>Remote Cluster security mode not supported</td>
<td>Occurs during EMCC login when the visiting cluster security mode is in mixed mode, the phone is in secure mode, and the home cluster is in nonsecure mode.</td>
</tr>
<tr>
<td>47</td>
<td>DN has multiple users(47)</td>
<td>DN has multiple users</td>
<td>Occurs during EMCC login when the Extension used for login is assigned for multiple users as Primary.</td>
</tr>
</tbody>
</table>
Extension Mobility Roaming Across Clusters Overview

Extension Mobility Roaming Across Clusters gives users the ability to roam across multiple clusters and make or receive calls even when the user's home cluster is down. This feature leverages the Intercluster Lookup Service (ILS) to replicate Extension Mobility users' directory numbers across all the clusters.

When a user logs into a roaming cluster, their phone registers to the roaming cluster using the directory number. Unlike Extension Mobility Cross Cluster (EMCC), where the phone from the visiting cluster registers to the home cluster, the roaming feature allows the user to maintain their registration in whichever cluster they are visiting.

Configuration Overview

To deploy this feature, you must do the following:

- Set up an ILS network - ILS is used to synchronize directory number across the clusters.


- Set up a uniform dial plan - you require a uniform dial plan across the ILS network.

- Device profile and user information must be synced in all the clusters.
- Configure Extension Mobility.
- Configure Roaming access to your Extension Mobility users.
- Super User Synchronization Tool - to synchronize the superuser information across the clusters.

**System Requirements for Extension Mobility Roaming Across Clusters**

The following system requirements exist for Cisco Unified Communications Manager:

- Cisco Unified Communications Manager, Release 12.0(1)SU1 or higher.
- Cisco Extension Mobility service must be running.
- Intercluster Lookup Service must be running.

**Extension Mobility Roaming Across Clusters Login**

**Login Terminology**

The following figure depicts the home cluster versus roaming cluster in Extension Mobility Roaming across Cluster.

*Figure 7: Home Cluster vs. Roaming Cluster*

**Home Cluster**

Home cluster is a cluster, where the user configuration such as User Device profile, Dial Plans reside here.

**Roaming Cluster**

Roaming cluster is a cluster, where users can do the Extension Mobility login to any Extension Mobility capable phone just like in their home cluster.
Superuser

A superuser is a user, who is associated to the Standard EM Roaming Across Clusters Super Users access control group. This user has a privilege to do the Extension Mobility login from a roaming cluster and can make or receive calls.

**Note**

- Superuser information must be shared across all the clusters irrespective of the cluster in which user is logging-in.
- A web-based application called **Super User Synchronization Tool** is provided to synchronize the superuser information across the clusters. This sync happens during a scheduled sync time that admin can configure on the tool. You can download the **Super User Synchronization Tool** from the [software.cisco.com](http://software.cisco.com).

Login Process

Cisco Unified Communications Manager supports the Extension Mobility login for a superuser created across multiple clusters. Extension Mobility login, in the roaming cluster allows superuser to access their phone settings, such as line appearances, services, dial plans. A superuser can make or receive calls from the roaming cluster, in the same way as they do in the home cluster.

*Figure 8: Call Flow When a User Is in Home Cluster*

In the preceding figure, let us assume Bob's DN as 1000-001, Alice's DN as 2000-001 and Kally's DN as 3000-001 registered with Cluster 1, 2 and 3 respectively. When Kally dials Bob's DN 1000-001, Cluster 3 routes the call to Cluster 1 and Bob and Kally are connected.
Let us assume Bob's home cluster is down and Bob is configured as superuser who can roam across the clusters. When Bob moves to Cluster 2 and does Extension Mobility Login, hosting phone gets re-registered with Bob's settings. Once the login is successful, all other clusters are updated with Bob's new location. Now when Kally dials Bob's DN 1000-01, Cluster 3 routes the call to Cluster 2 and Bob and Kally are connected. Similarly, Bob can call Kally by dialing DN 3000-001.

**Note**

- If a superuser did the Extension Mobility login to another cluster, user will automatically log out from the home cluster. If the cluster is down, it waits until the cluster is up to log out from the user's previous login.
- Extension Mobility Roaming Across Clusters supports the multi login behavior. Hence, superuser can login from multiple devices within the same cluster but not across the clusters.
**ILS Interaction**

In Cisco Unified CM Administration, you can configure ILS on a pair of clusters and then join those clusters to form an ILS network. Once you have established the ILS network, you can join additional clusters to the network without having to configure the connections between each cluster.

Whenever Extension Mobility login or logout occurs, ILS sync starts to update the available information to other clusters.

---

**Note**

Configuring user as superuser automatically initiates the ILS sync irrespective of Directory Number configuration for ILS.


---

**Extension Mobility Roaming Across Clusters 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>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that support the Extension Mobility feature.</td>
</tr>
</tbody>
</table>
| **Step 2** | Configure Extension Mobility by completing, the following subtasks in order:  
  - Activate Extension Mobility Services, on page 394  
  - Configure the Cisco Extension Mobility Phone Service, on page 395  
  - Create an Extension Mobility Device Profile for Users, on page 396  
  - Associate a Device Profile to a User, on page 402  
  - Subscribe to Extension Mobility, on page 402 | Configure Extension Mobility to allow users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones when they login from remote cluster. Perform this task flow on both home and remote clusters, so that users will be able to access settings from either a home or remote cluster. |
| **Step 3** | Configure Roaming for Extension Mobility Users, on page 460 | Use this procedure to give Extension Mobility users the ability to roam between different clusters in an ILS network, while using the same login credentials. |
**Generate a Phone Feature List**

Generate a phone feature list report to determine which devices support the feature that you want to configure.

**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>System Reports</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>From the list of reports, click <strong>Unified CM Phone Feature List</strong>.</td>
</tr>
</tbody>
</table>
| **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. |

**Activate Extension Mobility Services**

**Before you begin**

Generate a Phone Feature List, on page 1

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified Serviceability, choose <strong>Tools &gt; Service Activation</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>From the <strong>Server</strong> drop-down list, choose the required node.</td>
</tr>
</tbody>
</table>
| **Step 3** | Activate the following services:  
  a) Cisco CallManager  
  b) Cisco Tftp  
  c) Cisco Extension Mobility  
  d) ILS Service  
  **Note** You must choose publisher node to activate the ILS services. |
| **Step 4** | Click **Save**. |
| **Step 5** | Click **OK**. |

**What to do next**

Configure the Cisco Extension Mobility Phone Service, on page 395
Configure the Cisco Extension Mobility Phone Service

Configure the extension mobility IP phone service to which users can later subscribe to access extension mobility.

Before you begin

Activate Extension Mobility Services, on page 394

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; Device Settings &gt; Phone Services.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Service Name field, enter a name for the service.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Service URL field, enter the Service URL. The format is http://&lt;IP Address&gt;:8080/emapp/EMAppServlet?device=#DEVICENAME#. IP Address is the IP address of the Cisco Unified Communications Manager where Cisco Extension Mobility is activated and running. It can either be a IPv4 or a IPv6 address.</td>
</tr>
</tbody>
</table>

Example:

http://123.45.67.89:8080/emapp/EMAppServlet?device=#DEVICENAME#

Example:


This format allows a user to sign-in using User ID and PIN. You can configure more sign-in options for IP phone users who have subscribed to the extension mobility service. To configure more sign-in options, append the loginType parameter to the Service URL, in the following formats:

- **loginType=DN** enables users to sign in using Primary Extension and PIN.
  - The Service URL format is: http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=DN.

- **loginType=SP** enables users to sign in using Self Service UserID and PIN.
  - The Service URL format is: http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=SP.

- **loginType=UID** enables users to sign in using UserID and PIN.
  - The Service URL format is: http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=UID.

If you do not append loginType to the end of the URL, the default sign-in option displayed is User ID and PIN.

| Step 5 | In the Service Type field, choose whether the service is provisioned to the Services, Directories, or Messages button. |
Create an Extension Mobility Device Profile for Users

Configure an extension mobility device profile. This profile acts as a virtual device that maps onto a physical device when a user logs in to extension mobility. The physical device takes on the characteristics in this profile.

Before you begin
Configure the Cisco Extension Mobility Phone Service, on page 395

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 choose an existing device profile from the resulting list.
• To add a new device profile, click Add New, choose an option from the Device Profile Type, and click Next. Then, choose a device protocol from the Device Protocol drop-down list and click Next.

Step 3 Configure the fields. See the Related Topics section for more information about the fields and their configuration options.
Step 4 Click Save.
Step 5 From the Association Information section, click Add a new DN.
Step 6 In the Directory Number field, enter the directory number and click Save.
Step 7 Click Reset and follow the prompts.

What to do next
Associate a Device Profile to a User, on page 402

Related Topics
Device Profile Fields for Extension Mobility, on page 397

Associate a Device Profile to a User

Associate a device profile to users so that they can access their settings from a different phone. You associate a user device profile to a user in the same way that you associate a physical device.
You can use the Bulk Administration Tool (BAT) to add and delete several user device profiles for Cisco Extension Mobility at one time. 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.

**Tip**

**Before you begin**

Create an Extension Mobility Device Profile for Users, on page 396

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose **User Management** > **End User**.

**Step 2**  Perform one of the following tasks:

- To modify the settings for an existing user, enter search criteria, click **Find**, and choosing an existing user from the resulting list.
- To add a new user, click **Add New**.

**Step 3**  Under **Extension Mobility**, locate the device profile that you created and move it from **Available Profiles** to **Controlled Profiles**.

**Step 4**  Check the **Home Cluster** check box.

**Step 5**  Click **Save**.

**What to do next**

Subscribe to Extension Mobility, on page 402

**Subscribe to Extension Mobility**

Subscribe IP phones and device profiles to the extension mobility service so that users can log in, use, and log out of extension mobility.

**Before you begin**

Associate a Device Profile to a User, on page 402

**Procedure**

**Step 1**  Perform one of the following tasks from Cisco Unified CM Administration:

- Choose **Device** > **Phone**, specify search criteria, click **Find**, and choose a phone which users will use for extension mobility.
- Choose **Device** > **Device Settings** > **Device Profile**, specify search criteria, click **Find**, and choose the device profile that you created.
Configure Roaming for Extension Mobility Users

Use this procedure to give Extension Mobility users the ability to roam between different clusters in an ILS network, while using the same login credentials. To do this, you must assign the selected users to the Standard EM Roaming Across Clusters Super Users access control group.

Before you begin
An ILS network must have been set up as ILS is used to replicate user and login information across the clusters.

Procedure

Step 1  From Cisco Unified CM Administration, choose User Management > User Settings > Access Control Group.
Step 2  Click Find and select the Standard EM Roaming Across Clusters Super Users group.
Step 3  Click Add End Users to Group button. The Find and List Users pop-up window appears.
Step 4  Click Find and select all the users to whom you want to provide roaming ability.
Step 5  Click Add Selected.

Extension Mobility Roaming Across Clusters Interactions and Restrictions

Extension Mobility Roaming Across Clusters Interactions
This section lists the interactions of the Extension Mobility Roaming across Cluster with other Cisco Unified Communications Manager Administration components.

- Extension Mobility
- Inter-Cluster Lookup Service (ILS)

Extension Mobility Roaming Across Clusters Restrictions
This section lists the restrictions of the Extension Mobility Roaming across Cluster with other Cisco Unified Communications Manager Administration components.

- If the hub ILS is down, the spokes connected to it will not synchronize until the hub is back.
Different Types of Extension Mobility

The following table lists the different types of Extension Mobility features available in Cisco Unified Communications Manager and their differences.

Table 42: Differences Between EM, EMCC, and Extension Mobility Roaming Across Cluster

<table>
<thead>
<tr>
<th>Description</th>
<th>Extension Mobility (EM)</th>
<th>Extension Mobility Cross Cluster (EMCC)</th>
<th>Extension Mobility Roaming Across Cluster</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allows users to temporarily access their phone settings from other phones in the same cluster.</td>
<td>Allows users to access their phone settings from a phone in another cluster.</td>
<td>Allows user to roam across other clusters using own login credentials.</td>
<td></td>
</tr>
<tr>
<td>When the user logs in to a phone in another cluster</td>
<td>N/A</td>
<td>The remote cluster phone registers to the user’s home cluster, accessing the settings in the home cluster.</td>
<td>The roaming cluster phone registers in the roaming cluster only.</td>
</tr>
<tr>
<td>Intercluster</td>
<td>Single cluster only</td>
<td>Multiple clusters</td>
<td>Multiple clusters</td>
</tr>
<tr>
<td>Configuration</td>
<td>Single cluster only</td>
<td>EMCC must be configured in the home cluster and each cluster that the user visit.</td>
<td>Extension Mobility Roaming must be configured in all the cluster.</td>
</tr>
<tr>
<td>User Information</td>
<td>Single cluster only</td>
<td>Must be maintained in all clusters.</td>
<td>Superuser information maintained in all the cluster.</td>
</tr>
</tbody>
</table>

Extension Mobility Roaming Across Clusters Troubleshooting

This section provides information about error codes for EMApp and EMService.

Authentication Error

Problem “Error 201 Authentication Error” appears on the phone.

Solution The user should check that the correct user ID and PIN were entered; the user should check with the system administrator that the user ID and PIN are correct.

Blank User ID or PIN

Problem “Error 202 Blank User ID or PIN” appears on the phone.

Solution Enter a valid user ID and PIN.
Busy Please Try Again

Problem "Error 26 Busy Please Try Again" appears on the phone.

Solution Check whether the number of concurrent login and logout requests is greater than the Maximum Concurrent requests service parameter. If so, lower the number of concurrent requests.

Note To verify the number of concurrent login and logout requests, use the Cisco Unified Real-Time Monitoring Tool to view the Requests In Progress counter in the Extension Mobility object. For more information, see the Cisco Unified Real-Time Monitoring Tool Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html

Database Error

Problem "Error 6 Database Error” appears on the phone.

Solution Check whether a large number of requests exists. If a large number of requests exists, the Requests In Progress counter in the Extension Mobility object counter shows a high value. If the requests are rejected because of a large number of concurrent requests, the Requests Throttled counter also shows a high value. Collect detailed database logs.

Dev Logon Disabled

Problem "Error 22 Dev Logon Disabled” appears on the phone.

Solution Verify that you checked the Enable Extension Mobility check box in the Phone Configuration window (Device > Phone).

Device Name Empty

Problem “Error 207 Device Name Empty” appears on the phone.

Solution Check that the URL that is configured for Cisco Extension Mobility is correct. See the Related Topics section for more information.

Related Topics Configure the Cisco Extension Mobility Phone Service, on page 395

EM Service Connection Error

Problem “Error 207 EM Service Connection Error” appears on the phone.

Solution Verify that the Cisco Extension Mobility service is running by selecting Tools > Control Center—Feature in Cisco Unified Serviceability.

Host Not Found

Problem The “Host Not Found” error message appears on the phone.
**Solution** Check that the Cisco Tomcat service is running by selecting Tools > Control Center—Network Services in Cisco Unified Serviceability.

**HTTP Error**

**Problem** HTTP Error (503) appears on the phone.

**Solution**

- If you get this error when you press the Services button, check that the Cisco IP Phone Services service is running by selecting Tools > Control Center—Network Services in Cisco Unified Serviceability.
- If you get this error when you select Extension Mobility service, check that the Cisco Extension Mobility Application service is running by selecting Tools > Control Center—Network Services in Cisco Unified Serviceability.

**Phone Resets**

**Problem** After users log in or log out, their phones reset instead of restarting.

**Possible Cause** Locale change is the probable cause of the reset.

**Solution** No action is required. If the user locale that is associated with the logged-in user or profile is not the same as the locale or device, after a successful login the phone will restart and then reset. This pattern occurs because the phone configuration file is rebuilt.

**Phone Services Unavailable After Login**

**Problem** After logging in, the user finds that the phone services are not available.

**Possible Cause** This problem occurs because the user profile had no services associated with it when it was loaded on the phone.

**Solution**

- Ensure that the user profile includes the Cisco Extension Mobility service.
- Change the configuration of the phone where the user is logged in to include Cisco Extension Mobility. After the phone is updated, the user can access the phone services.

**Phone Services Unavailable After Logout**

**Problem** After a user logs out and the phone reverts to the default device profile, the phone services are no longer available.

**Solution**

- Verify that the Synchronization Between Auto Device Profile and Phone Configuration enterprise parameter is set to True.
- Subscribe the phone to the Cisco Extension Mobility service.
User Logged in Elsewhere

**Problem** “Error 25 User Logged in Elsewhere” appears on the phone.

**Solution** Check whether the user is logged in to another phone. If multiple logins must be allowed, ensure that the Multiple Login Behavior service parameter is set to Multiple Logins Allowed.

User Profile Absent

**Problem** “Error 205 User Profile Absent” appears on the phone.

**Solution** Associate a device profile to the user.
Hold Reversion

• Hold Reversion Overview, on page 465
• Hold Reversion Prerequisites, on page 465
• Hold Reversion Configuration Task Flow, on page 466
• Hold Reversion Interactions and Restrictions, on page 469

Hold Reversion Overview

When you place a call on hold, the Hold Reversion feature alerts you when the held call exceeds a configured time limit. When the configured time limit expires, an alert is generated on your phone to remind you to handle the call.

The following alerts are available:

• The Phone rings or beeps once
• The status line displays “Hold Reversion”
• The LED next to the line button flashes continuously
• A vibrating handset icon displays

The type of alert that you receive depends on the capabilities of your phone.

To retrieve a reverted call, you can:

• Pick up the handset
• Press the speaker button on the phone
• Press the headset button
• Select the line that is associated with the reverted call
• Press the Resume softkey

For details, see the user guide for your particular phone model.

Hold Reversion Prerequisites

• Cisco CallManager service must be running on at least one node in the cluster
• Cisco CTIManager service must be running on at least one node in the cluster
Hold Reversion Configuration Task Flow

Perform the following steps to configure Hold Reversion for your phones. This procedure assumes that you have configured directory numbers for phones, or that you are using auto-registration.

Before you begin

- If phone users want the hold reversion messages to display in a language other than English, or if you want the user to receive country-specific tones for calls, verify that you have installed the locale installer.
- Review Hold Reversion Prerequisites, on page 465

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Generate a Phone Feature List, on page 1</td>
<td>Run a phone feature list report to determine which phones support the Hold Reversion feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Call Focus Priority for Hold Reversion, on page 466</td>
<td>Configure the call focus priority setting against the device pool for your phones.</td>
</tr>
</tbody>
</table>
| Step 3 | Perform one of the following procedures:  
  • Configure Hold Reversion Timer Defaults for Cluster, on page 467  
  • Configure Hold Reversion Timer Settings for Phone, on page 468 | Configure the Hold Reversion timer settings. You can configure the timer using a clusterwide service parameter, or configure the settings on an individual phone line.  
**Note** The settings on an individual phone line override the clusterwide service parameter settings. |

Configure Call Focus Priority for Hold Reversion

As an administrator, you can prioritize incoming calls and reverted calls. By default, all incoming calls are handled before reverted calls, however you can change the call focus priority so that reverted calls take precedence.

Before you begin

Generate a Phone Feature List, on page 1
Procedure

Step 1 From Cisco Unified CM Administration, choose System > Device Pool and open the device pool that applies to your phones.

Step 2 In the Reverted Call Focus Priority field, choose one of the following options and click Save:
   • Default—Incoming calls have priority over reverted calls.
   • Highest—Reverted calls have priority over incoming calls.

Step 3 Click Save.

Step 4 Reset any devices in the Device Pool by performing the following steps:
   a) Click Reset. The Device Reset window displays.
   b) In the Device Reset window, click Reset.

What to do next
Perform one of the following procedures to configure Hold Reversion Timer Settings:
   • Configure Hold Reversion Timer Defaults for Cluster, on page 467
   • Configure Hold Reversion Timer Settings for Phone, on page 468

Configure Hold Reversion Timer Defaults for Cluster

Perform this procedure to configure clusterwide service parameters that apply hold reversion timer default settings for all phones in the cluster.

Note
When you configure the clusterwide service parameters, the configuration is applied as the default hold reversion setting for all phones in the cluster. However, the settings on an individual phone line can override the clusterwide defaults.

Before you begin
Configure Call Focus Priority for Hold Reversion, on page 466

Procedure

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

Step 2 From the Server drop-down list box, choose the server that is running the CallManager service.

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

Step 4 Configure values for the following clusterwide service parameters:
   • Hold Reversion Duration—Enter a number from 0 to 1200 (inclusive) to specify the wait time in seconds before Cisco Unified Communications Manager issues a reverted call alert to the holding party phone.
If you enter 0, Cisco Unified Communications Manager does not issue reverted call alerts, unless it is configured on a phone line.

- **Hold Reversion Interval Notification**—Enter a number from 0 to 1200 (inclusive) to specify the wait time in seconds before Cisco Unified Communications Manager sends periodic reminder alerts to the holding party phone. If you enter 0, Cisco Unified Communications Manager does not send periodic reminder alerts unless the timer is configured on a phone line.

**Step 5**  
Click Save.

---

**What to do next**

Perform the following procedure to configure the Hold Reversion timer for an individual phone:

Configure Hold Reversion Timer Settings for Phone, on page 468

---

**Configure Hold Reversion Timer Settings for Phone**

Perform this procedure to configure Hold Reversion timer settings for a phone and phone line.

**Note**
You can also configure Hold Reversion timer settings using a clusterwide service parameter. However, the settings on an individual phone line override the clusterwide service parameter setting.

**Before you begin**

Perform Configure Hold Reversion Timer Defaults for Cluster, on page 467 to configure Hold Reversion clusterwide defaults.

**Procedure**

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

**Step 2**  
Click Find and select the phone on which you want to configure Hold Reversion.

**Step 3**  
In the Association pane on the left, click the phone line on which you want to configure Hold Reversion.

**Step 4**  
Configure values for the following fields:

- **Hold Reversion Ring Duration**—Enter a number from 0 to 1200 (inclusive) to specify the wait time in seconds before Cisco Unified Communications Manager issues a reverted call alert. If you enter 0, Cisco Unified Communications Manager does not issue reverted call alerts to this DN. If you leave the field empty (the default setting), Cisco Unified Communications Manager applies the setting from the Hold Reversion Duration service parameter.

- **Hold Reversion Ring Interval Notification**—Enter a number from 0 to 1200 (inclusive) to specify the wait time in seconds before Cisco Unified Communications Manager sends periodic reminder alerts. If you enter 0, Cisco Unified Communications Manager does not send periodic reminder alerts to this DN. If you leave the field empty (the default setting), Cisco Unified Communications Manager applies the setting from the Hold Reversion Interval Notification service parameter.

**Step 5**  
Click Save.
Step 6  
Reset the phone by performing the following steps:

a)  Click **Reset**. The **Reset Device** window displays.

b)  Click **Reset**.

Hold Reversion Interactions and Restrictions

Hold Reversion Interactions

**Table 43: Hold Reversion Feature Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Music on Hold</td>
<td>MOH is supported on a reverted call if MOH is configured for a normal held call.</td>
</tr>
<tr>
<td>Call Park</td>
<td>If hold reversion is invoked and the held party presses the <strong>Park</strong> softkey, the holding party still receives hold reversion alerts and can retrieve the call. When the holding party retrieves the call, the holding party receives MOH, if configured. If the held party parks before the hold duration exceeds the configured time limit, the system suppresses all hold reversion alerts until the call is picked up or redirected.</td>
</tr>
<tr>
<td>MLPP</td>
<td>When a multilevel precedence and preemption (MLPP) call is put on hold and reverts, the MLPP call loses its preemption status, and the reverted call gets treated as a routine call. After the call reverts, the system does not play a preemption ring. If a high precedence call becomes a reverted call, the system does not play a precedence tone.</td>
</tr>
</tbody>
</table>
| CTI Applications              | CTI applications can access hold reversion functionality when the feature is enabled for a line or the system. Cisco-provided applications such as Cisco Unified Communications Manager Assistant and attendant console provide hold reversion functionality using the CTI interface. When hold reversion gets invoked, the CTI port receives event notification instead of the audible tone presented on Cisco Unified IP Phones. CTI ports and route points receive the event notification once only, whereas Cisco Unified IP Phones receive alerts at regular intervals. See the following API documents for information about CTI requirements and interactions with hold reversion:
  - *Cisco Unified Communications JTAPI Developer Guide*
  - *Cisco Unified Communications TAPI Developer Guide* |
| Hold Reversion Interval for SCCP phones when interacting with SIP Phones | SCCP phones support a minimum Hold Reversion Notification Interval (HRNI) of 5 seconds, whereas SIP phones support a minimum of 10 seconds. SCCP phones set for the minimum HRNI of 5 seconds may experience a Hold Reversion Notification ring delay of 10 seconds when handling calls involving SIP phones. |
Hold Reversion Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Shared Lines</strong></td>
<td>If a Cisco Unified IP Phone that supports hold reversion shares a line with a phone device that does not support hold reversion, the hold reversion configuration settings display only for the line on the supporting device.</td>
</tr>
<tr>
<td></td>
<td>If a shared line device disables the feature, hold reversion gets disabled on all other devices that share the line.</td>
</tr>
<tr>
<td><strong>Ring Settings</strong></td>
<td>If the ring settings that are configured for the phone specify Disabled, the phone does not ring, flash, or beep for the hold reversion feature.</td>
</tr>
</tbody>
</table>

---

## Hold Reversion Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco Extension Mobility and Cisco Web Dialer</strong></td>
<td>Cisco Extension Mobility and Cisco Web Dialer features do not support the hold reversion feature.</td>
</tr>
<tr>
<td><strong>Directory numbers</strong></td>
<td>If a directory number is associated to a phone that does not support hold reversion, the feature settings do not display for that directory number in the Directory Number Configuration window.</td>
</tr>
<tr>
<td><strong>Shared lines</strong></td>
<td>If a Cisco Unified IP Phone that supports hold reversion shares a line with a phone device that does not support hold reversion, the hold reversion configuration settings display only for the line on the supporting device.</td>
</tr>
<tr>
<td></td>
<td>If a shared-line device disables this feature, hold reversion gets disabled on all other devices that share this line.</td>
</tr>
<tr>
<td><strong>Ring settings</strong></td>
<td>Hold reversion ring uses the ring settings that Cisco Unified Communications Manager Administration defines for that user (disable, flash only, ring once, ring, beep only) except that flash gets converted to flash once, and ring gets converted to ring once.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>When an IP Phone call is on normal hold, the ring settings (Phone Idle) from the Call Manager is applied.</td>
</tr>
<tr>
<td><strong>Maximum number of reverted calls</strong></td>
<td>The maximum number of reverted calls on a line equals the maximum number of calls on your system.</td>
</tr>
<tr>
<td><strong>Cisco Unified IP Phones</strong></td>
<td>You cannot configure hold reversion settings for DNs that are associated with phones that do not support this feature. Only Cisco Unified IP Phones that support the hold reversion feature display the hold reversion timer settings in the Directory Number Configuration window.</td>
</tr>
<tr>
<td></td>
<td>When Hold Reversion is configured for the system, the phone must support the feature or the feature does not activate.</td>
</tr>
<tr>
<td></td>
<td>See Cisco Unified IP Phone administration guides for Cisco Unified IP Phone models that support hold reversion and this version of Unified Communications Manager for any phone restrictions with hold reversion.</td>
</tr>
</tbody>
</table>
Accessing Hunt Groups

• Hunt Group Overview, on page 471
• Hunt Group Prerequisites, on page 472
• Hunt Group Configuration Task Flow, on page 472
• Hunt Group Interactions and Restrictions, on page 477

Hunt Group Overview

A Hunt Group is a group of lines that are organized hierarchically, so that if the first number in the hunt group list is busy, the system dials the second number. If the second number is busy, the system dials the next number, and so on.

The phone users can log in to or log out of the hunt groups by using the HLog softkey or the Hunt Group line button on the IP phone. The phone provides a visual status of the login state, so that the user can determine whether they are logged in to one or more of their line groups.

The Hunt Group feature provides the following functions:

• The HLog softkey on the IP phone allows the user to toggle between login and logout of phone.
• A hunt group allows a caller to automatically find an available line from amongst a group of extensions.
• The Hunt Group Log Off feature allows phone users to prevent their phones from receiving incoming calls that get routed to directory numbers. Regardless of the phone status, the phone rings normally for incoming calls that are not calls to one or more line groups associated with the phone.

Note

The directory numbers (DNs) belong to line groups that are associated with the phone.

• System administrators can log in or log out the users from the phones that are automatically logged into hunt groups.
• The HLog softkey allows a phone user to log a phone out of all line groups to which the phone directory numbers belong.
• From Cisco Unified Communications Manager Release 9.0 onward, the Hunt Group Log Off feature enables the use of mobile device as a desk phone. When you use the Hlog softkey through your mobile client, you no longer receive calls that are placed to the hunt pilot.
Hunt Group Prerequisites

- The phones must be running Skinny Client Control Protocol (SCCP) or Session Initiation Protocol (SIP).
- The phone ringtone file must be located in the TFTP directory (/usr/local/cm/tftp).

Hunt Group Configuration Task Flow

Before you begin

- Review Hunt Group Prerequisites, on page 472

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 Softkey Template for Hunt Group, on page 472</td>
<td>Configure a softkey template for the HLog softkey.</td>
</tr>
</tbody>
</table>
| Step 2 | To Associate a Softkey Template with a Common Device Configuration, on page 474, complete the following subtasks:  
  • Add a Softkey Template to a Common Device Configuration, on page 474  
  • Associate a Common Device Configuration with a Phone, on page 475 | Optional. 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. |
| Step 3 | Associate a Softkey Template with a Phone, on page 475 | 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 to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment. |
| Step 4 | Configure Phones for Hunt Group, on page 476 | Configure phones to automatically log in to or log out of hunt groups and hunt lists. |

Configure a Softkey Template for Hunt Group

The HLog softkey appears on the phone when the phone is in the following call states:

- Connected
• On Hook
• Off Hook

Note
You must create a new softkey template to configure the HLog softkey. You cannot configure the HLog softkey in a standard softkey template.

Use this procedure to make the HLog softkey available:

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:

• If you modified a template that is already associated with devices, click Apply Config 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 procedures:

- Add a Softkey Template to a Common Device Configuration, on page 474
- Associate a Softkey Template with a Phone, on page 475

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 Associate a Softkey Template with a Phone, on page 475.

Before you begin

Configure a Softkey Template for Hunt Group, on page 472

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add a Softkey Template to a Common Device Configuration, on page 474</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Common Device Configuration with a Phone, on page 475</td>
</tr>
</tbody>
</table>

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.
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**

**Before you begin**

**Add a Softkey Template to a Common Device Configuration**, on page 474

**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.

**Associate a Softkey Template with a Phone**

This procedure is optional. You can 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: 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.

**Before you begin**

**Configure a Softkey Template for Hunt Group**, on page 472

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Device > Phone**.
The **Find and List Phones** window appears.
Configure Phones for Hunt Group

Use this procedure to configure phones to automatically log in to or log out of hunt groups and hunt lists.

Before you begin

Ensure the phone directory numbers belong to one or more hunt groups.

See the Cisco Unified Communications Manager Administration Guide for information on hunt groups and hunt lists.

Procedure

Step 1
In the Cisco Unified CM Administration, choose Device > Phone.

Step 2
Perform one of the following tasks:

a) To modify the fields for an existing phone, enter search criteria and choose a phone from the resulting list. The Phone Configuration window appears.

b) To add a new phone, click Add New.

   The Add a New Phone window appears.

Step 3
In the Phone Configuration window, perform one of the following tasks:

a) To log out the phone from the hunt group, uncheck the Log into Hunt Group check box.

b) To log in to the phone to the hunt group, ensure that the Log into Hunt Group check box is checked.

   Note  
   The Log into Hunt Group check box remains checked by default for all phones.

Step 4
Click Save.

Related Topics

Configure Hunt Group Service Parameter, on page 476

Configure Hunt Group Service Parameter

The Hunt Group Logoff Notification service parameter provides the option to turn audible ringtones on or off when calls that come in to a line group arrive at a phone that is currently logged out. This ringtone alerts a logged-out user that there is an incoming call to a hunt list to which the line is a member, but the call will not ring at the phone of that line group member because of the logged-out status.

To configure the Hunt Group Logoff Notification service parameter, perform the following steps.
Procedure

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

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

Step 3  From the Service drop-down list, choose Cisco CallManager.
        The Service Parameter Configuration window appears.

Step 4  In the Clusterwide Parameters (Device - Phone) section, configure values for the following Hunt Group Logoff Notification service parameter:
        Enter a name for the ringtone file that Cisco IP Phones play when a member of a line group (hunt group) has logged out. The default value for this service parameter is None, which indicates no ringtone. You can enter a maximum of 255 characters.

Step 5  Click Save.
        The window refreshes, and Cisco Unified Communications Manager updates the service parameter with your changes.

Hunt Group Interactions and Restrictions

Hunt Group Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Non-shared-line Directory Number</td>
<td>If a phone is logged out of a line group and an extension on the phone is not shared, the line group does not ring that directory number (DN) in the line group. When the line group would normally offer the call to the DN, call processing skips the DN and acts as if the DN does not belong to the line group.</td>
</tr>
</tbody>
</table>
| Shared-line Directory Number  | Because the Log Out of Hunt Group feature is device-based, when a user logs a phone out, the feature affects only the logged-out phone. Calls to a line group that contains a shared-line directory number behave as follows:  
  • The DN does not ring if all phones that share that DN are logged out.  
  • The DN does ring if one or more phones that share the DN are logged in.  
  • The audible ring on a phone that is logged out is turned off by default. Cisco Unified Communications Manager provides a system parameter that can be set, so that a different ring tone plays when a call comes in to a logged-out hunt group member. |
## Hunt Group Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Multiple Line Groups</strong></td>
<td>When the user enables the Hunt Group Log Off feature by pressing the HLog softkey, the phone gets logged out from all associated line groups. This is because Hunt Group Log Off is a device-based feature. If a phone has DN's that belong to multiple line groups, pressing the HLog softkey logs the phone out of all associated line groups.</td>
</tr>
</tbody>
</table>
| **7940, 7960, and third-party SIP phones** | • When a phone that is running SIP (7906, 7911, 7941, 7961, ) is logged in to hunt groups and Call Forward All is activated, the call gets presented to the phone that is running SIP.  
• When 7940 and 7960 phones that are running SIP are logged in to hunt groups and Call Forward All is activated, the phones get skipped and the next phone in the line group rings.  
• 7940 and 7960 phones that are running SIP and third-party phones that are running SIP can be logged in to or logged out of hunt groups by using the **Phone Configuration** window, but no softkey support exists.  
• 7940 and 7960 phones that are running SIP and third-party phones that are running SIP do not show “Logged out of hunt groups” on the status line.  
• 7940 and 7960 phones that are running SIP and third-party phones that are running SIP do not play the Hunt Group Logoff Notification tone regardless of whether the tone is configured. |
Malicious Call Identification Overview

You can configure the Malicious Call Identification (MCID) feature to track troublesome or threatening calls. Users can report these calls by requesting that Cisco Unified Communications Manager identify and register the source of the incoming call in the network.

When the MCID feature is configured, the following actions take place:

1. The user receives a threatening call and presses Malicious call (or enters the feature code *39 if using a POTS phone that is connected to an SCCP gateway).
2. Cisco Unified Communications Manager sends the user a confirmation tone and a text message, if the phone has a display, to acknowledge receiving the MCID notification.
3. Cisco Unified Communications Manager updates the call details record (CDR) for the call with an indication that the call is registered as a malicious call.
4. Cisco Unified Communications Manager generates the alarm and local syslogs entry that contains the event information.
5. Cisco Unified Communications Manager sends an MCID invocation through the facility message to the connected network. The facility information element (IE) encodes the MCID invocation.
6. After receiving this notification, the PSTN or other connected network can take actions, such as providing legal authorities with the call information.

Malicious Call Identification Prerequisites

- Gateways and connections that support MCID:
  - PRI gateways that use the MGCP PRI backhaul interface for T1 (NI2) and E1 (ETSI) connections
  - H.323 trunks and gateways
- IP Phones that support MCID
## Malicious Call Identification Configuration Task Flow

**Before you begin**

- Review *Malicious Call Identification Prerequisites*, on page 479

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that support the MCID feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Set Malicious Call ID Service Parameter, on page 481</td>
<td>Enable Cisco Unified Communications Manager to flag a call detail record (CDR) with the MCID indicator.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Malicious Call ID Alarms, on page 481</td>
<td>Configure alarms to ensure that alarm information displays in the system logs.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure a Softkey Template for Malicious Call Identification, on page 482</td>
<td>Configure a softkey template with MCID. <strong>Note</strong> The Cisco Unified IP Phones 8900 and 9900 Series support MCID with feature button only.</td>
</tr>
</tbody>
</table>
| Step 5 | To Associate a Softkey Template with a Common Device Configuration, on page 483, complete the following subtasks:  
  - Add a Softkey Template to a Common Device Configuration, on page 483  
  - Associate a Common Device Configuration with a Phone, on page 484 | Optional. 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. |
| Step 6 | Associate a Softkey Template with a Phone, on page 484 | 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. |
| Step 7 | To Configure Malicious Call Identification Button, on page 485, complete the following subtasks:  
  - Configure Malicious Call ID Phone Button Template, on page 485 | Perform this step to add and configure the MCID button to a phone. |
Set Malicious Call ID Service Parameter

To enable Cisco Unified Communications Manager to flag a CDR with the MCID indicator, you must enable the CDR flag.

**Before you begin**

Configure Malicious Call ID Alarms, on page 481

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Step Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose System &gt; Service Parameters.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list, choose the Cisco Unified Communications Manager server name.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Service drop-down list, choose Cisco CallManager. The Service Parameter Configuration window displays.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the System area, set the CDR Enabled Flag field to True.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

**What to do next**

Configure Malicious Call ID Alarms, on page 481

Configure Malicious Call ID Alarms

In the Local Syslogs, you must set the alarm event level and activate alarms for MCID.

**Before you begin**

Set Malicious Call ID Service Parameter, on page 481

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Step Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified Serviceability, choose Alarm &gt; Configuration. The Alarm Configuration window displays.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list, choose the Cisco Unified Communications Manager server and click Go.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Service Group drop-down list, choose CM Services. The Alarm Configuration window updates with configuration fields.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Service drop-down list, choose Cisco CallManager.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Under Local Syslogs, in the Alarm Event Level drop-down list, choose Informational. The Alarm Configuration window updates with configuration fields.</td>
</tr>
</tbody>
</table>
**Configure a Softkey Template for Malicious Call Identification**

**Note**
Skinny Client Control Protocol (SCCP) IP phones use a softkey to invoke the MCID feature.

---

**Before you begin**
Configure Malicious Call ID Alarms, on page 481

**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**
In the Select a call state to configure field, choose Connected.
The list of Unselected Softkeys changes to display the available softkeys for this call state.

**Step 7**
In the Unselected Softkeys drop-down list, choose Toggle Malicious Call Trace (MCID).

**Step 8**
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 9 Click Save.

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 Associate a Softkey Template with a Phone, on page 484.

Before you begin
Configure a Softkey Template for Malicious Call Identification, on page 482

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add a Softkey Template to a Common Device Configuration, on page 483</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Common Device Configuration with a Phone, on page 484</td>
</tr>
</tbody>
</table>

Add a Softkey Template to a Common Device Configuration

Before you begin
Configure a Softkey Template for Malicious Call Identification, on page 482

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.
**Associate a Common Device Configuration with a Phone**

**Before you begin**

*Add a Softkey Template to a Common Device Configuration, on page 483*

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Find** and select the phone device to add the softkey template.
3. From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.
4. Click **Save**.
5. Click **Reset** to update the phone settings.

**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**

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Find** to select the phone to add the softkey template.
3. From the **Softkey Template** drop-down list, choose the template that contains the new softkey.
4. Click **Save**.
Configure Malicious Call Identification Button

The procedures in this section describe how to configure the Malicious Call Identification button.

Before you begin

Configure Malicious Call ID Alarms, on page 481

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Malicious Call ID Phone Button Template, on page 485.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Button Template with a Phone, on page 486</td>
</tr>
</tbody>
</table>

Configure Malicious Call ID Phone Button Template

Before you begin

Configure Malicious Call ID Alarms, on page 481

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:

- If you modified a template that is already associated with devices, click **Apply Config** to restart the devices.
• If you created a new softkey template, associate the template with the devices and then restart them.

**What to do next**

*Associate a Button Template with a Phone*, on page 486

### Associate a Button Template with a Phone

**Before you begin**

*Configure Malicious Call ID Phone Button Template*, on page 485

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Action Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified CM Administration, choose <strong>Device &gt; Phone</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click <strong>Find</strong> to display the list of configured phones.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Choose the phone to which you want to add the phone button template.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>In the <strong>Phone Button Template</strong> drop-down list, choose the phone button template that contains the new feature button.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click <strong>Save</strong>. A dialog box is displayed with a message to press <strong>Reset</strong> to update the phone settings.</td>
</tr>
</tbody>
</table>

---

### Malicious Call Identification Interactions and Restrictions

#### Malicious Call Identification Interactions

**Table 44: Malicious Call Identification Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference Calls</td>
<td>When a user is connected to a conference, the user can use the MCID feature to flag the call as a malicious call. Cisco Unified Communications Manager sends the MCID indication to the user, generates the alarm, and updates the CDR. However, Cisco Unified Communications Manager does not send an MCID invoke message to the connected network that might be involved in the conference.</td>
</tr>
<tr>
<td>Extension Mobility</td>
<td>Extension Mobility users can have the MCID softkey as part of their user device profile and can use this feature when they are logged on to a phone.</td>
</tr>
<tr>
<td>Call Detail Records</td>
<td>To track malicious calls by using CDR, you must set the <strong>CDR Enabled Flag</strong> to <strong>True</strong> in the Cisco CallManager service parameter. When the MCID feature is used during a call, the CDR for the call contains <strong>CallFlag=MALICIOUS</strong> in the Comment field.</td>
</tr>
</tbody>
</table>
To record alarms for the MCID feature in the Local Syslogs, you must configure alarms in Cisco Unified Serviceability. Under Local Syslogs, enable alarms for the Informational alarm event level.

When the MCID feature is used during a call, the system logs an SDL trace and a Cisco Unified Communications Manager trace in alarms. You can view the Alarm Event Log by using Cisco Unified Serviceability. The traces provide the following information:

- Date and time
- Type of event: Information
- Information: The Malicious Call Identification feature is invoked in Cisco Unified Communications Manager
- Called Party Number
- Called Device Name
- Called Display Name
- Calling Party Number
- Calling Device Name
- Calling Display Name
- Application ID
- Cluster ID
- Node ID


The Cisco ATA 186 analog phone ports support MCID by using the feature code (*39).

### Malicious Call Identification Restrictions

**Table 45: Malicious Call Identification Restrictions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
</table>
| Alarms  | To record alarms for the MCID feature in the Local Syslogs, you must configure alarms in Cisco Unified Serviceability. Under Local Syslogs, enable alarms for the Informational alarm event level. When the MCID feature is used during a call, the system logs an SDL trace and a Cisco Unified Communications Manager trace in alarms. You can view the Alarm Event Log by using Cisco Unified Serviceability. The traces provide the following information:  
- Date and time  
- Type of event: Information  
- Information: The Malicious Call Identification feature is invoked in Cisco Unified Communications Manager  
- Called Party Number  
- Called Device Name  
- Called Display Name  
- Calling Party Number  
- Calling Device Name  
- Calling Display Name  
- Application ID  
- Cluster ID  
- Node ID  
| Cisco ATA 186 analog phone ports | The Cisco ATA 186 analog phone ports support MCID by using the feature code (*39). |

Malicious Call Identification Terminating (MCID-T) function  
Cisco Unified Communications Manager supports only the malicious call identification originating function (MCID-O). Cisco Unified Communications Manager does not support the malicious call identification terminating function (MCID-T). If Cisco Unified Communications Manager receives a notification from the network of a malicious call identification, Cisco Unified Communications Manager ignores the notification.
<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercluster trunks</td>
<td>MCID does not work across intercluster trunks because Cisco Unified Communications Manager does not support the MCID-T function.</td>
</tr>
<tr>
<td>Cisco MGCP FXS gateways</td>
<td>Cisco MGCP FXS gateways do not support MCID. No mechanism exists for accepting the hookflash and collecting the feature code in MGCP.</td>
</tr>
<tr>
<td>QSIG trunks</td>
<td>MCID does not work over QSIG trunks because MCID is not a QSIG standard.</td>
</tr>
<tr>
<td>Cisco VG248 Analog Phone Gateway</td>
<td>Cisco VG248 Analog Phone Gateway does not support MCID.</td>
</tr>
<tr>
<td>SIP trunks</td>
<td>MCID does not support SIP trunks.</td>
</tr>
<tr>
<td>Immediate Divert</td>
<td>System does not support using MCID and Immediate Divert features together.</td>
</tr>
</tbody>
</table>

**Malicious Call ID Troubleshooting**

To track and troubleshoot Malicious Call ID, you can use Cisco Unified Communications Manager SDL traces and alarms. For information about setting traps and traces for MCID, see the *Cisco Unified Serviceability Administration Guide*. For information about how to generate reports for MCID, see the *Cisco Unified CDR Analysis and Reporting Administration Guide*. 
Call Transfer

Call Transfer Overview

The transfer feature allows you to redirect a connected call from your phone to another number. After call transfer, your call is disconnected and the transferred call is established as a new call connection.

Following are the different types of call transfers:

- **Consult Transfer and Blind Transfer**—In Consult Transfer, a transferring phone user can redirect the caller to a different target address, after consulting with the target phone user that answers the call. That is, the transferring phone user will stay on the call until the target phone user answers the call. In Blind Transfer, the transferring phone user connects the caller to a destination line before the target of the transfer answers the call.

  Most phones use hard keys or softkeys for Transfer. Both Consult Transfer and Blind Transfer do not require separate configuration. The difference between the two types of transfer depends on when the transferring party presses the Transfer button a second time. For a consult transfer, the transferring party presses the Transfer button after the target answers, while for a Blind Transfer, the transferring party presses the Transfer button before the target answers.

  For SCCP-initiated blind transfers, Cisco Unified Communications Manager provides call progress indications in the form of ring-back to the transferred user.

- **Transfer On-Hook**—In this type of call transfer, the user presses the Transfer softkey, dials the number to which the call will be transferred, and then presses the Transfer softkey again, or simply goes on-hook to complete the transfer operation. You must set the Transfer On-Hook parameter to True. This service parameter determines whether a call transfer is completed as a result of the user going on-hook after initiating a transfer operation.

  Both Consult Transfer and Blind Transfer use the Transfer On-Hook option.

- **Direct Transfer**—This type of transfer allows a user to join two established calls (the two calls can either be on hold or in the connected state) into one call and then drop the initiator from the transfer. Direct Transfer does not initiate a consultation call and does not put the active call on hold. The user uses the DirTrfr softkey to join any two established calls and remove the initiator.
## Call Transfer 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 Consult and Blind Transfer, on page 490</td>
<td>Transfer allows you to redirect a single call to a new number with or without consulting the transfer recipient. Perform this step to configure Trnsfr as a softkey and/or button.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Transfer On-Hook, on page 495</td>
<td>(Optional) Transfer On-Hook is an option to complete call transfers. Press Trnsfr, dial the number to which the call should be transferred to, and go on-hook to complete the transfer. Perform this step to configure the service parameter.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure Direct Transfer, on page 495</td>
<td>(Optional) Direct Transfer allows you to transfer two calls to each other (without you remaining on the line). Perform this step to configure DirTrfr as a softkey and/or button.</td>
</tr>
</tbody>
</table>

## Configure Consult and Blind Transfer

Complete one of the task flows depending on whether your phone supports softkey or buttons.

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure a Softkey Template for Transfer, on page 490</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Transfer Button, on page 494</td>
<td></td>
</tr>
</tbody>
</table>

## Configure a Softkey Template for Transfer

Transfer softkey is used for consult and blind transfer of a call. The transfer softkey has the following call states:

- **Connected**
- **On Hold**

Use this procedure to make the Transfer softkey available:

### 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:

   • If you modified a template that is already associated with devices, click **Apply Config** 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 procedures:

- **Associate Transfer Softkey Template with a Common Device Configuration**, on page 491
- **Associate Transfer Softkey Template with a Phone**, on page 493

### Associate Transfer 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 Associate Transfer Softkey Template with a Phone, on page 493.

Before you begin
Configure a Softkey Template for Transfer, on page 490

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add Transfer Softkey Template to the Common Device Configuration, on page 492</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Common Device Configuration with a Phone, on page 493</td>
</tr>
</tbody>
</table>

What to do next
Configure Transfer Button, on page 494

Add Transfer Softkey Template to the Common Device Configuration

Before you begin
Configure a Softkey Template for Transfer, on page 490

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 493

---

**Associate a Common Device Configuration with a Phone**

**Before you begin**

**Add Transfer Softkey Template to the Common Device Configuration**, on page 492

**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 <strong>Device &gt; Phone</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> and select the phone device to add the softkey template.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Common Device Configuration</strong> drop-down list, choose the common device configuration that contains the new softkey template.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Reset</strong> to update the phone settings.</td>
</tr>
</tbody>
</table>

---

**Associate Transfer 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.

**Before you begin**

**Configure a Softkey Template for Transfer**, on page 490

**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 <strong>Device &gt; Phone</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> to select the phone to add the softkey template.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Softkey Template</strong> drop-down list, choose the template that contains the new softkey.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Press <strong>Reset</strong> to update the phone settings.</td>
</tr>
</tbody>
</table>
Configure Transfer Button

The procedures in this section describe how to configure the Transfer button.

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure a Phone Button Template for Transfer, on page 494</td>
<td>Perform this step to assign Transfer button features to line or speed dial keys.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate Transfer Button Template with a Phone, on page 495</td>
<td>Perform this step to configure the Transfer button for a phone.</td>
</tr>
</tbody>
</table>

### Configure a Phone Button Template for Transfer

Optional. Follow this procedure when you want to assign features to line or speed dial keys.

#### 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:
  - If you modified a template that is already associated with devices, click Apply Config to restart the devices.
  - If you created a new softkey template, associate the template with the devices and then restart them.

### What to do next

Associate Transfer Button Template with a Phone, on page 495
**Associate Transfer Button Template with a Phone**

**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 Device &gt; Phone.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find to display the list of configured phones.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose the phone to which you want to add the phone button template.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Phone Button Template drop-down list, choose the phone button template that contains the new feature button.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save. A dialog box is displayed with a message to press Reset to update the phone settings.</td>
</tr>
</tbody>
</table>

**Configure Transfer On-Hook**

**Before you begin**

Configure Consult and Blind Transfer, on page 490

**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; Service Parameters. The Service Parameter Configuration window is displayed.</td>
</tr>
<tr>
<td>Step 2</td>
<td>From the Server drop-down list, choose the server on which you want to configure the parameter.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Service drop-down list, choose the Cisco CallManager (Active) service.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Clusterwide Parameters (Device - Phone), choose True for the Transfer On-Hook Enabled service parameter.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

**What to do next**

(Optional) Configure Direct Transfer, on page 495

**Configure Direct Transfer**

Complete one of the task flows depending on whether your phone supports softkey or buttons.

**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 Softkey Template for Direct Transfer, on page 496</td>
<td>Perform this step to add Direct Transfer softkey to template and configure the softkey using the Common Device Configuration or phone.</td>
</tr>
</tbody>
</table>
## Configure a Softkey Template for Direct Transfer

Direct Transfer softkey has the following call states:

- Connected
- On hold

Use this procedure to make the Direct Transfer softkey available:

### 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:

- If you modified a template that is already associated with devices, click **Apply Config** 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 the following procedures:

• Associate Direct Transfer Softkey Template with a Common Device Configuration, on page 497

• Associate Direct Transfer Softkey Template with a Phone, on page 498

Associate Direct Transfer 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 Associate Direct Transfer Softkey Template with a Phone, on page 498

Before you begin
Configure a Softkey Template for Direct Transfer, on page 496

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Add Direct Transfer Softkey Template to the Common Device Configuration, on page 497</td>
<td>Perform this step to add Direct Transfer softkey template to the Common Device Configuration.</td>
</tr>
<tr>
<td>Step 2 Associate a Common Device Configuration with a Phone, on page 498</td>
<td>Perform this step to add Direct Transfer softkey template to the Common Device Configuration.</td>
</tr>
</tbody>
</table>

Add Direct Transfer Softkey Template to the 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.

---

### What to do next

**Associate a Common Device Configuration with a Phone**, on page 498

---

**Associate a Common Device Configuration with a Phone**

### Before you begin

**Add Direct Transfer Softkey Template to the Common Device Configuration**, on page 497

### 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.

---

**Associate Direct Transfer 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.

### Before you begin

**Configure a Softkey Template for Direct Transfer**, on page 496
Configure Direct Transfer Button

The procedures in this section describe how to configure the Direct Transfer button.

<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>Configure Phone Button Template for Direct Transfer, on page 499</strong></td>
<td>Perform this step to assign Direct Transfer button features to line or speed dial keys.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>Associate Direct Transfer Button Template with a Phone, on page 500</strong></td>
<td>Perform this step to configure the Direct Transfer button for a phone.</td>
</tr>
</tbody>
</table>

Configure Phone Button Template for Direct Transfer

*Optional.* Follow this procedure when you want to assign features to line or speed dial keys.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; Phone Button Template.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Find to display list of supported phone templates.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.</td>
</tr>
<tr>
<td></td>
<td>a) Select a default template for the model of phone and click Copy.</td>
</tr>
<tr>
<td></td>
<td>b) In the Phone Button Template Information field, enter a new name for the template.</td>
</tr>
<tr>
<td></td>
<td>c) Click Save.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Perform the following steps if you want to add phone buttons to an existing template.</td>
</tr>
<tr>
<td></td>
<td>a) Click Find and enter the search criteria.</td>
</tr>
<tr>
<td></td>
<td>b) Choose an existing template.</td>
</tr>
<tr>
<td>Step 5</td>
<td>From the Line drop-down list, choose feature that you want to add to the template.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Perform one of the following tasks:</td>
</tr>
<tr>
<td></td>
<td>• If you modified a template that is already associated with devices, click Apply Config to restart the devices.</td>
</tr>
</tbody>
</table>
If you created a new softkey template, associate the template with the devices and then restart them.

What to do next

Associate Direct Transfer Button Template with a Phone, on page 500

Associate Direct Transfer Button Template with a Phone

Before you begin

Configure Phone Button Template for Direct Transfer, on page 499

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.
## Call Transfer Interactions and Restrictions

### Call Transfer Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Logical Partitioning</td>
<td>The logical partitioning policy check is performed between the geolocation identifier of the device that is acting as a transferred party and the geolocation identifier of the device that is acting as a transferred destination.</td>
</tr>
</tbody>
</table>

Logical partitioning handling takes place in the following circumstances:

- When a phone user uses Transfer softkey to transfer the call, the second press of the softkey invokes and processes the Call Transfer feature.
- When other transfer mechanisms, such as Direct Transfer, On-Hook Transfer, Hook Flash Transfer, and CTI-application-initiated Transfer results in invoking the Call Transfer feature.
- When the transferred and the transferred destination specifies a PSTN participant.
- When Cisco Unified Communications Manager uses the geolocation identifier information that associates with the transferred and transferred destination device to perform logical partitioning policy checking.
- Before splitting of the primary and secondary calls, and before joining.

Logical partitioning handles a denied call as follows:

- Sends External Transfer Restricted message to the VoIP phone.
- Normal Transfer—For a phone that is running SCCP, the primary call remains on hold, and the consultation call remains active. For a phone that is running SIP, both primary and consultation calls remain on hold and must be resumed manually after the failure.
- On-Hook, Hook-Flash and Analog-Phone-Initiated Transfer—Both the primary and secondary calls are cleared by using the cause code=63 “Service or option not available” with a reorder tone from Cisco Unified Communications Manager.
- The Number of Transfer Failures perfmon counter is incremented.
Interaction

When a switch initiates a call transfer between two segments that have the same precedence level, the segments maintain the precedence level upon transfer. When a call transfer is made between call segments that are at different precedence levels, the switch that initiates the transfer marks the connection at the segment that has the higher precedence level.

Cisco Unified Communications Manager supports this requirement by upgrading the precedence level of a call leg that is involved in a Call Transfer operation. For example, party A calls party B with Priority precedence level. Party B then initiates a transfer to party C and dials the Flash precedence digits when dialing. When the transfer is complete, the precedence level of party A gets upgraded from Priority to Flash.

The Call Transfer feature is enabled automatically when MLPP is enabled, and the phones support the Transfer softkey.

Note: The precedence level upgrade does not work over a trunk device such as an intercluster trunk (ICT) or a PRI trunk.

Call Transfer Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Logical Partitioning</td>
<td>Logical partitioning handling does not take place when both the transferred and the transferred destination devices are VoIP phones. Logical partitioning handling does not take place when geolocation or a geolocation filter is not associated with any device.</td>
</tr>
<tr>
<td>External Call Transfer Restrictions</td>
<td>To restrict transfer for external call scenarios, see the “External Call Transfer Restrictions” chapter.</td>
</tr>
<tr>
<td>Hunt Pilot</td>
<td>If a call transfer to a hunt pilot is initiated when an announcement is in progress, the call is redirected only after the announcement is complete.</td>
</tr>
</tbody>
</table>
External Call Transfer Restrictions

External Call Transfer Restrictions Overview

External Call Transfer Restrictions is a feature that you can use to configure gateways, trunks, and route patterns as OnNet (internal) or OffNet (external) devices at the system level. By setting the devices as OffNet, you can restrict the transferring of an external call to an external device and thus help prevent toll fraud.

If you try to transfer a call on an OffNet gateway or trunk when the service parameter Block OffNet to OffNet Transfer is set to True, a message displays on the user phone to indicate that the call cannot be transferred.

This chapter uses the following terms:

<table>
<thead>
<tr>
<th>Term</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>OnNet Device</td>
<td>A device that is configured as OnNet and considered to be internal to the network.</td>
</tr>
<tr>
<td>OffNet Device</td>
<td>A device that is considered as OffNet and, when routed, is considered to be external to the network.</td>
</tr>
<tr>
<td>Network Location</td>
<td>The location of the device, which is considered as OnNet or OffNet, with respect to the network.</td>
</tr>
<tr>
<td>Originating End</td>
<td>The device that gets transferred. The system considers this device as OnNet or OffNet.</td>
</tr>
<tr>
<td>Terminating End</td>
<td>The device that receives the transferred call. The system considers this device as OnNet or OffNet.</td>
</tr>
<tr>
<td>Incoming Call</td>
<td>A call for which only gateways and trunks call classification settings get used to classify it as OnNet or OffNet. Route Pattern call classification settings do not apply.</td>
</tr>
</tbody>
</table>
Configure External Call Transfer Restrictions Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure the Service Parameter for Call Transfer Restrictions, on page 504</td>
<td>Block external calls from being transferred to another external device or number.</td>
</tr>
<tr>
<td><strong>Step 2</strong> To configure incoming calls perform the following procedures:</td>
<td>Configure gateways and trunks as OnNet (internal) or OffNet (external) by using Gateway Configuration or Trunk Configuration or by setting a clusterwide service parameter.</td>
</tr>
<tr>
<td>• Configure the Clusterwide Service Parameter, on page 505</td>
<td></td>
</tr>
<tr>
<td>• Configure Gateways for Call Transfer Restrictions, on page 506</td>
<td></td>
</tr>
<tr>
<td>• Configure Trunks for Call Transfer Restrictions, on page 507</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure Outgoing Calls, on page 507</td>
<td>Configure transfer capabilities with route pattern configuration.</td>
</tr>
</tbody>
</table>

**Configure the Service Parameter for Call Transfer Restrictions**

To block external calls from being transferred to another external device or number:

**Procedure**

1. From the Cisco Unified CM Administration user interface choose **System > Service Parameters**.
2. On the Service Parameter Configuration window choose the Cisco Unified CM server you want to configure from the Server drop-down menu.
3. Choose **Cisco CallManager (Active)** from the Service drop-down menu.
4. Choose **True** from the Block OffNet to OffNet Transfer drop-down menu. The default value specifies False.
5. Click **Save**.
Configure the Clusterwide Service Parameter

To configure all gateways or trunks in the Cisco Unified Communications Manager cluster to be OffNet (external) or OnNet (internal), perform the following steps:

### Configure Incoming Calls 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 all gateways or trunks in the Cisco Unified Communications Manager cluster to be OffNet (external) or OnNet (internal).</td>
</tr>
<tr>
<td>(Optional) Configure the Clusterwide Service Parameter, on page 505</td>
<td></td>
</tr>
</tbody>
</table>
| **Step 2**        | Configure gateways as OnNet (internal) or OffNet (external) by using Gateway Configuration. When the feature is used in conjunction with the clusterwide service parameter Block OffNet to OffNet Transfer, the configuration determines whether calls can transfer over a gateway. You can configure the following devices as internal and external to Cisco Unified Communications Manager:  
  - H.323 gateway  
  - MGCP FXO trunk  
  - MGCP T1/E1 trunk |
| Configure Gateways for Call Transfer Restrictions, on page 506 |
| **Step 3**        | Configure trunks as OnNet (internal) or OffNet (external) by using Trunk Configuration. When the feature is used in conjunction with the clusterwide service parameter Block OffNet to OffNet Transfer, the configuration determines whether calls can transfer over a trunk. You can configure the following devices as internal and external to Cisco Unified Communications Manager:  
  - Intercluster trunk  
  - SIP trunk |
| Configure Trunks for Call Transfer Restrictions, on page 507 |
**Before you begin**

Configure the Service Parameter for Call Transfer Restrictions, on page 504

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From the Cisco Unified CM Administration user interface choose <strong>System &gt; Service Parameters</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>On the Service Parameter Configuration window choose the Cisco Unified CM server you want to configure from the Server drop-down menu.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose <strong>Cisco CallManager (Active)</strong> from the Service drop-down menu.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Choose either OffNet or OnNet (the default specifies OffNet) from the Call Classification drop-down menu.</td>
</tr>
</tbody>
</table>

**What to do next**

Configure Gateways for Call Transfer Restrictions, on page 506

---

**Configure Gateways for Call Transfer Restrictions**

To configure the gateway as OffNet, OnNet, or Use System Default, perform the following procedure. The system considers calls that come to the network through that gateway as OffNet or OnNet, respectively.

**Before you begin**

Configure the Clusterwide Service Parameter, on page 505

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified Communications Manager Administration, choose <strong>Device &gt; Gateway</strong>. The Find and List Gateways window displays.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>To list the configured gateways, click <strong>Find</strong>. The gateways that are configured in Cisco Unified Communications Manager display.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose the gateway that you want to configure as OffNet or OnNet.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Call Classification field choose OffNet or OnNet. If you have enabled clusterwide restrictions an all gateways, configure each gateway to Use System Default (this reads the setting in the Call Classification service parameter and uses that setting for the gateway).</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

**What to do next**

Configure Trunks for Call Transfer Restrictions, on page 507
Configure Trunks for Call Transfer Restrictions

To configure the trunk as OffNet, OnNet, or Use System Default, perform the following procedure. The system considers calls that come to the network through that trunk as OffNet or OnNet, respectively.

**Before you begin**

Configure Gateways for Call Transfer Restrictions, on page 506

**Procedure**

**Step 1**

From Cisco Unified Communications Manager Administration, choose **Device > Trunk**.

The Find and List Trunk window displays.

**Step 2**

To list the configured trunks, click **Find**.

The trunks that are configured in Cisco Unified Communications Manager display.

**Step 3**

Choose the trunk that you want to configure as OffNet or OnNet.

**Step 4**

From the Call Classification drop-down list, choose one of the following fields:

- **OffNet** - When you choose this field, this identifies the gateway as an external gateway. When a call comes in from a gateway that is configured as OffNet, the system sends the outside ring to the destination device.

- **OnNet** - When you choose this field, this identifies the gateway as an internal gateway. When a call comes in from a gateway that is configured as OnNet, the system sends the inside ring to the destination device.

- **Use System Default** - When you choose this field, this uses the Cisco Unified Communications Manager clusterwide service parameter Call Classification.

**Note**

If you have enabled clusterwide restrictions an all trunks, configure each trunk to Use System Default (this reads the setting in the Call Classification service parameter and uses that setting for the trunk)

**Step 5**

Click **Save**.

**What to do next**

Configure Outgoing Calls, on page 507

---

**Configure Outgoing Calls**

To classify a call as OnNet or OffNet, administrators can set the **Call Classification** field to OnNet or OffNet, respectively, on the **Route Pattern Configuration** window. Administrators can override the route pattern setting and use the trunk or gateway setting by checking the **Allow Device Override** check box on the **Route Pattern Configuration** window.
Before you begin
Configure Trunks for Call Transfer Restrictions, on page 507

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Route Pattern and click Find to list all route patterns.

Step 2 Choose the route pattern you want to configure, or click Add New.

Step 3 In the Route Pattern Configuration window, use the following fields to configure transfer capabilities with route pattern configuration:
   a) Call Classification—Use this drop-down list to classify the call that uses this route Pattern as OffNet or OnNet.
   b) Provide Outside Dial Tone—If Call Classification is set to OffNet, this check box gets checked.
   c) Allow Device Override—When this check box is checked, the system uses the Call Classification setting of the trunk or gateway that is associated with the route pattern instead of the Call Classification setting on the Route Pattern Configuration window.

Step 4 Click Save.

External Call Transfer Interactions and Restrictions

External Call Transfer Restrictions Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Drop Conference</td>
<td>The Drop Conference feature determines whether an existing ad hoc conference should be dropped by checking whether the conference parties are configured as OffNet or OnNet. You use the service parameter Drop Ad Hoc Conference and choose the option When No OnNet Parties Remain in the Conference to configure the feature. You determine OnNet status for each party by checking the device or route pattern that the party is using. For more information, see topics related to Ad Hoc Conference linking in the “Ad Hoc Conferencing” chapter.</td>
</tr>
<tr>
<td>Bulk Administration</td>
<td>Bulk Administration inserts gateway configuration (OffNet or OnNet) on the Gateway Template. For more information, see the Cisco Unified Communications Manager Bulk Administration Guide.</td>
</tr>
</tbody>
</table>
Interaction Feature

Dialed Number Analyzer (DNA)  When used to perform digit analysis on a gateway, DNA displays the Call Classification that is configured for the gateway and the route pattern. For more information, see the Cisco Unified Communications Manager Dialed Number Analyzer Guide.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dialed Number Analyzer (DNA)</td>
<td>When used to perform digit analysis on a gateway, DNA displays the Call Classification that is configured for the gateway and the route pattern. For more information, see the Cisco Unified Communications Manager Dialed Number Analyzer Guide.</td>
</tr>
</tbody>
</table>
PART X

Presence and Privacy Features

- Barge, on page 513
- BLF Presence, on page 525
- Call Display Restrictions, on page 541
- Do Not Disturb, on page 553
- Privacy, on page 565
- Private Line Automatic Ringdown, on page 571
- Secure Tone, on page 577
Barge Overview

Barge allows a user to be added to a remotely active call that is on a shared line. Remotely active calls for a line are the active (connected) calls that are made to or from another device that shares a directory number with the line.

If you configure party entrance tone, a tone plays on the phone when a basic call changes to a barged call or cbarge call. In addition, a different tone plays when a party leaves the multiparty call.

Phones support Barge in the following conference modes:

- Built-in conference bridge at the phone that is barged—This mode uses the Barge softkey. Most Cisco Unified IP Phones include the built-in conference bridge capability.
- Shared conference bridge—This mode uses the cBarge softkey.

By pressing the Barge or cBarge softkey in the remote-in-use call state, the user is added to the call with all parties, and all parties receive a barge beep tone (if configured). If Barge fails, the original call remains active.

If no conference bridge is available (built-in or shared), the barge request gets rejected, and a message displays on the Barge initiator device. When network or Cisco Unified Communications Manager failure occurs, the Barge call is preserved.

For a list of Cisco Unified IP Phones that support Barge, log in to Cisco Unified Reporting and run the Unified CM Phone Feature List report. Make sure to select Built In Bridge as the feature. For details, see Generate a Phone Feature List, on page 1.

Single-Button Barge and Single-Button cBarge

The Single-Button Barge and Single-Button cBarge features allow a user to press the shared-line button of the remotely active call, to be added to the call. All parties receive a barge beep tone (if configured). If barge fails, the original call remains active.

Phones support Single-Button Barge and Single-Button cBarge in two conference modes:

- Built-in conference bridge at the phone that is barged—This mode uses the Single-Button Barge feature.
Shared conference bridge—This mode uses the Single-Button cBarge feature.

By pressing the shared-line button of the remote-in-use call, the user is added to the call with all parties, and all parties receive a barge beep tone (if configured). If barge fails, the original call remains active. If no conference bridge is available (built-in or shared), the barge request gets rejected, and a message is displayed at the Barge initiator device.

**Built-In Conference**

When the user presses the Barge softkey or a shared-line button, a Barge call is set up by using the built-in conference bridge, if available. A built-in conference bridge is advantageous because neither a media interruption nor display changes to the original call occur when the Barge is being set up.

**Shared Conference**

When the user presses the cBarge softkey, or a shared-line button, a barge call is set up by using the shared conference bridge, if available. The original call is split and then joined at the conference bridge, which causes a brief media interruption. The call information for all parties changes to “Barge”. The barged call becomes a conference call with the barge target device as the conference controller. It can add more parties to the conference or can drop any party. When any party releases the call, the remaining two parties experience a brief interruption and then get reconnected as a point-to-point call, which releases the shared conference resource.

**Built-In and Shared Conference Differences**

This table describes the differences between barge with built-in conference bridge and shared conference.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Barge with Built-In Conference</th>
<th>Barge with Shared Conference</th>
</tr>
</thead>
<tbody>
<tr>
<td>The standard softkey template includes the Barge/cBarge softkey. Note: If the single button Barge/cBarge feature is enabled, the softkey is not used.</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>A media break occurs during barge setup.</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>If configured, a user receives a barge setup tone.</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Text displays at the barge initiator phone.</td>
<td>To barge XXX</td>
<td>To Conference</td>
</tr>
<tr>
<td>Text displays at the target phone.</td>
<td>To/From Other</td>
<td>To Conference</td>
</tr>
<tr>
<td>Text displays at the other phones.</td>
<td>To/From Target</td>
<td>To Conference</td>
</tr>
<tr>
<td>Bridge supports a second barge setup to an already barged call.</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Feature</td>
<td>Barge with Built-In Conference</td>
<td>Barge with Shared Conference</td>
</tr>
<tr>
<td>---------</td>
<td>-------------------------------</td>
<td>-------------------------------</td>
</tr>
<tr>
<td>Initiator releases the call.</td>
<td>No media interruption occurs for the two original parties.</td>
<td>Media break occurs to release the shared conference bridge when only two parties remain and to reconnect the remaining parties as a point-to-point call.</td>
</tr>
<tr>
<td>Target releases the call.</td>
<td>Media break occurs to reconnect initiator with the other party as a point-to-point call.</td>
<td>Media break occurs to release the shared conference bridge when only two parties remain and to reconnect the remaining parties as a point-to-point call.</td>
</tr>
<tr>
<td>Other party releases the call.</td>
<td>All three parties get released.</td>
<td>Media break occurs to release the shared conference bridge when only two parties remain and to reconnect the remaining parties as a point-to-point call.</td>
</tr>
<tr>
<td>Target puts call on hold and performs Direct Transfer, Join, or Call Park.</td>
<td>Initiator gets released.</td>
<td>Initiator and the other party remain connected.</td>
</tr>
</tbody>
</table>

**Barge Configuration Task Flow**

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th><strong>Command or Action</strong></th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Configure Softkey Template for Built-In Conferencing, on page 516</td>
<td>Add the Barge softkey to a softkey template. Follow this procedure when you are configuring barge for built-in conference bridges.</td>
</tr>
<tr>
<td></td>
<td>Configure Softkey Template for Shared Conferencing, on page 517</td>
<td>Add the cBarge softkey to a softkey template. Follow this procedure when you are configuring barge for shared conference bridges.</td>
</tr>
<tr>
<td></td>
<td>To Associate a Softkey Template with Common Device Configuration, on page 518, complete the following subtasks:</td>
<td>Optional. 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.</td>
</tr>
<tr>
<td></td>
<td>• Add a Softkey Template to Common Device Configuration, on page 519</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Associate Common Device Configuration with Phone, on page 519</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Associate Softkey Template with Phone, on page 518</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.</td>
</tr>
</tbody>
</table>
### Command or Action | Purpose
--- | ---
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.

**Step 5** | Configure Barge for Built-In Conferencing, on page 520
--- | ---
Configure barge for built-in conference bridges.

**Step 6** | Configure Barge for Shared Conferencing, on page 521
--- | ---
Configure barge for shared conference bridges.

**Step 7** | Associate User with Device, on page 61
--- | ---
Associate users with devices.

### Configure Softkey Template for Built-In Conferencing

Configure a softkey template for Barge and assign the Barge softkey to that template. You can configure the Barge softkey in the **Remote In Use** call state.

**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:
• If you modified a template that is already associated with devices, click **Apply Config** 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 procedures:

- **Add a Softkey Template to Common Device Configuration**, on page 519
- **Associate Common Device Configuration with Phone**, on page 519

**Configure Softkey Template for Shared Conferencing**

Configure a softkey template for shared conferencing and assign the cBarge softkey to that template. You can configure the cBarge softkey in the **Remote In Use** call state.

**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:
If you modified a template that is already associated with devices, click Apply Config 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.

## Associate Softkey Template with Phone

### Procedure

**Step 1** From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window is displayed.

**Step 2** Find the phone to which you want to add the softkey template.

**Step 3** Perform one of the following tasks:

- From the Common Device Configuration drop-down list, choose the common device configuration that contains the required softkey template.
- In the Softkey Template drop-down list, choose the softkey template that contains the Barge or cBarge softkey.

**Step 4** Click Save. A dialog box is displayed with a message to press Reset to update the phone settings.

## Associate a Softkey Template with 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 Associate Softkey Template with Phone, on page 518.

### Procedure

**Step 1** Add a Softkey Template to Common Device Configuration, on page 362

**Step 2** Associate a Common Device Configuration with a Phone, on page 362
Add a Softkey Template to Common Device Configuration

Before you begin
Perform one or both of the following as needed:
• Configure Softkey Template for Built-In Conferencing, on page 516
• Configure Softkey Template for Shared Conferencing, on page 517

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 Common Device Configuration with Phone, on page 519

Associate Common Device Configuration with Phone

Before you begin
Perform one or both of the following as needed:
• Configure Softkey Template for Built-In Conferencing, on page 516
• Configure Softkey Template for Shared Conferencing, on page 517
Configure Barge for Built-In Conferencing

Most Cisco Unified IP Phones include the built-in conference bridge capability; that is, these Cisco IP Phones have an internal DSP that acts as a small conference bridge to support the barge feature. It can support only a maximum of three parties that include the phone itself. Starting from firmware version 11.x, Cisco IP Phone 8800 Series have the capability to daisy chain the built-in bridge (BIB) feature.

### 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.

### What to do next

Perform one or both of the following:

- Configure Barge for Built-In Conferencing, on page 520
- Configure Barge for Shared Conferencing, on page 521

### Configure Barge for Built-In Conferencing

Most Cisco Unified IP Phones include the built-in conference bridge capability; that is, these Cisco IP Phones have an internal DSP that acts as a small conference bridge to support the barge feature. It can support only a maximum of three parties that include the phone itself. Starting from firmware version 11.x, Cisco IP Phone 8800 Series have the capability to daisy chain the built-in bridge (BIB) feature.

### Procedure

**Step 1** From Cisco Unified CM Administration, choose **System > Service Parameters** and set the **Built In Bridge Enable** clusterwide service parameter to **On**.

**Note** If this parameter is set to **Off**, configure barge for each phone by setting the **Built in Bridge** field in the **Phone Configuration** window.

**Step 2** Set the **Party Entrance Tone** clusterwide service parameter to **True** (required for tones) or configure the **Party Entrance Tone** field in the **Directory Number Configuration** window.

**Step 3** Set the **Single Button Barge/Call Barge Policy** to **Barge**.

**Note** If this parameter is set to **Off**, configure single-button barge for each phone by setting the **Single Button Barge** field in the **Phone Configuration** window.

**Step 4** Set the **Allow Barge When Ringing** service parameter to **True**.

**Step 5** Click **Save**.
Configure Barge for Shared Conferencing

Cisco recommends that you do not configure Barge for shared conferencing (cBarge) for a user who has Barge configured. Choose only one barge method for each user.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Service Parameters and set the Built In Bridge Enable clusterwide service parameter to On.

   Note If this parameter is set to Off, configure cBarge for each phone by setting the Built in Bridge field in the Phone Configuration window.

Step 2 Set the Party Entrance Tone clusterwide service parameter to True (required for tones) or configure the Party Entrance Tone field in the Directory Number Configuration window.

Step 3 Set the Single Button Barge/CBarge Policy to cBarge.

   Note If this parameter is set to Off, configure Single-button cBarge for each phone by setting the Single Button cBarge field in the Phone Configuration window.

Step 4 Set the Allow Barge When Ringing service parameter to True.

Step 5 Click Save.

Associate User with Device

Before you begin

Perform one or both of the following:

- Configure Barge for Built-In Conferencing, on page 520
- Configure Barge for Shared Conferencing, on page 521

Procedure

Step 1 From Cisco Unified CM Administration, choose User Management > End User.

Step 2 Specify the appropriate filters in the Find User Where field to and then click Find to retrieve a list of users.

Step 3 Select the user from the list.

The End User Configuration window appears.

Step 4 Locate the Device Information section.

Step 5 Click Device Association.

The User Device Association window appears.

Step 6 Find and select the CTI remote device.

Step 7 To complete the association, click Save Selected/Changes.

Step 8 From Related Links drop-down list box, choose Back to User, and then click Go.
The **End User Configuration** window appears, and the associated device that you chose appears in the **Controlled Devices** pane.

---

## Barge Interactions and Restrictions

This section describes the interactions and restrictions for barge and privacy features.

### Barge Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>cBarge</td>
<td>Cisco recommends that you assign either the Barge or cBarge softkey to a softkey template. By having only one of these softkeys for each device, you can prevent confusion for users and avoid potential performance issues.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> You can enable Single-Button Barge or Single-Button cBarge for a device, but not both.</td>
</tr>
<tr>
<td>Call Park</td>
<td>When the target parks the call, the barge initiator gets released (if using the built-in bridge), or the barge initiator and the other party remain connected (if using the shared conference).</td>
</tr>
<tr>
<td>Join</td>
<td>When the target joins the call with another call, the barge initiator gets released (if using the built-in bridge), or the barge initiator and the other party remain connected (if using the shared conference).</td>
</tr>
<tr>
<td>Private Line Automatic Ringdown (PLAR)</td>
<td>A Barge, cBarge, or Single-Button Barge initiator can barge into a call through a shared line that is configured for Barge and Private Line Automatic Ringdown (PLAR). The initiator can barge into the call if the barge target uses the preconfigured number that is associated with the PLAR line while on the call. Cisco Unified Communications Manager does not send the barge invocation to the PLAR line before connecting the barge call, so the barge occurs regardless of the state of the PLAR destination.</td>
</tr>
<tr>
<td></td>
<td>To make Barge, cBarge, or Single-Button Barge function with PLAR, you must configure Barge, cBarge, or Single-Button Barge. In addition, you must configure the PLAR destination, a directory number that is used specifically for PLAR.</td>
</tr>
</tbody>
</table>

### Barge Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Additional callers</td>
<td>The Barge initiator cannot conference in additional callers.</td>
</tr>
<tr>
<td>Computer Telephony Interface (CTI)</td>
<td>CTI does not support Barge through APIs that TAPI and JTAPI applications invoke. CTI generates events for Barge when it is invoked manually from an IP phone by using the Barge or cBarge softkey.</td>
</tr>
</tbody>
</table>
Barge Troubleshooting

No Conference Bridge Available

When the Barge softkey is pressed, the message No Conference Bridge Available is displayed on the IP phone.

The Built In Bridge field in the Phone Configuration window for the target phone is not set properly.

To resolve the problem, perform the following steps:

1. From Cisco Unified CM Administration, choose Device > Phone and click Find the phone to find the phone configuration of the phone that is having the problem.
2. Set the Built In Bridge field to On.
3. Click Update.
4. Reset the phone.

Error: Past Limit

The phone displays the message, Error: Past Limit.

The number of shared-line users in the conference is equal to or greater than the configuration for the Maximum Number of Calls field for the device from which you are attempting to barge.

- Go to Service Parameter Configuration window and locate the Clusterwide Parameters (Feature - Conference) section. Increase the value of Maximum Ad Hoc Conference parameter as required.
- Check the Maximum Number of Calls value for the shared lines on the device from which you are attempting to barge and increase the value as required.
BLF Presence

BLF Presence Overview

The Busy Lamp Field (BLF) presence feature allows a user who is a watcher to monitor the real-time status of another user at a directory number or Session Initiation Protocol (SIP) uniform resource identifier (URI) from the device of the watcher.

A watcher can monitor the status of the user or BLF presence entity (also called presentity) by using the following options:

- BLF and SpeedDial buttons
- Missed call, placed call, or received call lists in the directories window
- Shared directories, such as the corporate directory

Call lists and directories display the BLF status for existing entries. When you configure BLF and SpeedDial buttons, the BLF presence entity appears as a speed dial on the device of the watcher.

To view the status of a BLF presence entity, watchers send BLF presence requests to Cisco Unified Communications Manager. After administrators configure BLF presence features, real-time status icons appear on the watcher device to indicate whether the BLF presence entity is on the phone, is not on the phone, the status is unknown, and so on.

Extension mobility users can use BLF presence features on phones with extension mobility support.

BLF presence group authorization ensures that only authorized watchers can access the BLF presence status for a destination. Because the administrator ensures that the watcher is authorized to monitor the destination when a BLF or Speed Dial is configured, BLF presence group authorization does not apply to BLF or Speed Dials.

BLF Presence Prerequisites

- Configure the phones that you want to use with the BLF presence feature.
Configure the SIP trunks that you want to use with the BLF presence feature.

## BLF Presence Configuration Task Flow

### Before you begin

- Review **BLF Presence Prerequisites**, on page 525

### Procedure

| Step 1 | Configure and synchronize cluster-wide enterprise parameters for Busy Lamp Field (BLF). See **Configure/Synchronize Cluster-Wide Enterprise Parameters for BLF**, on page 527. | Configure BLF options that apply to all devices and services in the same cluster. You can synchronize enterprise-parameter configuration changes with the configured devices in the least-intrusive manner. For example, a reset or restart may not be required on some affected devices. |
| Step 2 | Configure cluster-wide service parameters for BLF. See **Configure Cluster-Wide Service Parameters for BLF**, on page 528. | Configure presence service parameters to configure different services on selected servers in Cisco Unified Communications Manager Administration. |
| Step 3 | Configure BLF presence groups. See **Configure BLF Presence Groups**, on page 528. | Configure BLF presence groups to control the destinations that watchers can monitor. |
| Step 4 | To associate BLF presence group with devices and users, perform the following subtasks:  
  - Associate BLF presence groups with phones. See **Associate BLF Presence Groups with Phone**, on page 530.  
  - Associate BLF presence groups with SIP trunks. See **Associate BLF Presence Groups with SIP Trunk**, on page 531.  
  - Associate BLF presence groups with an end user. See **Associate BLF Presence Groups with End User**, on page 532.  
  - Associate BLF presence groups with an application user. See **Associate BLF Presence Groups with Application User**, on page 533. | Apply a BLF presence group to a directory number, SIP trunk, phone that is running SIP, phone that is running SCCP, application user (for application users that are sending presence requests over the SIP trunk), or end user. |
| Step 5 | Accept BLF presence requests from external trunks and applications. See **Accept BLF Presence Requests from External Trunks and Applications**, on page 534. | To enable application-level authorization for a SIP trunk application in addition to trunk-level authorization. |
## Configure/Synchronize Cluster-Wide Enterprise Parameters for BLF

Use enterprise parameters for default configuration that apply to all devices and services in the same cluster. A cluster consists of a set of Cisco Unified Communications Managers that share the same database. When you install a new Cisco Unified Communications Manager, it uses the enterprise parameters to set the initial values of its device defaults.

### Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose System &gt; Enterprise Parameters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure the fields in the Enterprise Parameters Configuration window. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Tip</td>
<td>For details about an enterprise parameter, click the parameter name or the question mark that appears in the Enterprise Parameter Configuration window.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 4</td>
<td>(Optional) Click Apply Config to synchronize cluster-wide parameters. The Apply Configuration Information dialog box appears.</td>
</tr>
</tbody>
</table>

### Command or Action | Purpose |
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 6</strong></td>
<td>Configure Calling Search Space. See Configure a Calling Search Space for Presence Requests, on page 535.</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Configure a phone button template for BLF and SpeedDial buttons. See Configure a Phone Button Template for BLF and SpeedDial Buttons, on page 535.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Associate button template with a device. See Associate Button Template with a Device, on page 536.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>Configure user device profile. See Configure User Device Profile, on page 537.</td>
</tr>
</tbody>
</table>
Configure Cluster-Wide Service Parameters for BLF

You can configure one or multiple services available in the Service Parameter Configuration window for BLF.

Before you begin

Configure/Synchronize Cluster-Wide Enterprise Parameters for BLF, on page 527

Procedure

Step 1
In the 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 parameter.

Step 3
Configure the fields in the Service Parameters Configuration window. See the online help for more information about the fields and their configuration options.

Tip
For details about the service parameters, click the parameter name or the question mark that appears in the Service Parameter Configuration window.

Step 4
Click Save.

Note
The Default Inter-Presence Group Subscription parameter does not apply to BLF and SpeedDials.

What to do next

Configure BLF Presence Groups, on page 528

Configure BLF Presence Groups

You can use BLF presence groups to control the destinations that watchers can monitor. To configure a BLF presence group, create the group in Cisco Unified Communications Manager Administration and assign one or more destinations and watchers to the same group.

When you add a new BLF presence group, Cisco Unified Communications Manager defines all group relationships for the new group with the default cluster field as the initial permission fields. To apply different permissions, configure new permissions between the new group and existing groups for each permission that you want to change.

Note
The system always allows BLF presence requests within the same BLF presence group.
To view the status of a presence entity, watchers send presence requests to Cisco Unified Communications Manager. The system requires watchers to be authorized to initiate status requests for a presence entity with these requirements:

- The watcher BLF presence group be authorized to obtain the status for the presence entity presence group, whether inside or outside of the cluster.
- Unified CM must be authorized to accept BLF presence requests from an external presence server or application.

**Before you begin**

Configure Cluster-Wide Service Parameters for BLF, on page 528

**Procedure**

**Step 1**
In Cisco Unified CM Administration, choose **System > BLF Presence Group**.

**Step 2**
Configure the fields in the **BLF Presence Group Configuration** window. See the Related Topics section for details about the fields and their configuration options.

**Note**
Use the **Default Inter-Presence Group Subscription** service parameter for the Cisco CallManager service. It sets the clusterwide permissions parameter for BLF presence groups to allow subscription or disallow subscription. This field enables administrators to set a system default and configure BLF presence group relationships by using the default field for the cluster.

**Step 3**
Click **Save**.

**Note**
The permissions that you configure for a BLF presence group appear in the **BLF Presence Group Relationship** pane. Permissions that use the system default permission field for the group-to-group relationship do not appear.

**What to do next**

Associate BLF presence group with devices and users by performing the following subtasks:

- Associate BLF Presence Groups with Phone, on page 530
- Associate BLF Presence Groups with SIP Trunk, on page 531
- Associate BLF Presence Groups with End User, on page 532
- Associate BLF Presence Groups with Application User, on page 533

**Related Topics**

BLF Presence Group Fields for BLF, on page 529

**BLF Presence Group Fields for BLF**

Presence authorization works with BLF presence groups. The following table describes the BLF presence group configuration fields.
**BLF Presence Group Association with Devices and Users**

Perform the following procedures to apply a BLF presence group to the phone, SIP trunk, phone that is running SIP, phone that is running SCCP, directory number, application user (for application users that are sending presence requests over the SIP trunk), and end user.

*Note*  
The system allows presence requests between members in the same BLF presence group.

**Associate BLF Presence Groups with Phone**

You can use BLF presence for phones and trunks when the phones and trunks have permission to send and receive presence requests.

Cisco Unified Communications Manager handles the BLF presence requests for Cisco Unified Communications Manager users, whether inside or outside the cluster. For a Cisco Unified Communications Manager watcher that sends a BLF presence request through the phone, Cisco Unified Communications Manager responds with the BLF presence status if the phone and BLF presence entity are colocated.

---

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>Enter the name of the BLF presence group that you want to configure. For example, Executive_Group.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description for the BLF presence group that you are configuring.</td>
</tr>
<tr>
<td>Modify Relationship to Other Presence Groups</td>
<td>Select one or more BLF presence groups to configure the permission fields for the named group to the selected groups.</td>
</tr>
<tr>
<td>Subscription Permission</td>
<td>For the selected BLF presence groups, choose one of the following options from the drop-down list:</td>
</tr>
<tr>
<td></td>
<td>- <strong>Use System Default</strong>—Set the permissions field to the Default Inter-Presence Group Subscription clusterwide service parameter field (Allow Subscription or Disallow Subscription).</td>
</tr>
<tr>
<td></td>
<td>- <strong>Allow Subscription</strong>—Allow members in the named group to view the real-time status of members in the selected groups.</td>
</tr>
<tr>
<td></td>
<td>- <strong>Disallow Subscription</strong>—Block members in the named group from viewing the real-time status of members in the selected groups.</td>
</tr>
</tbody>
</table>

The permissions that you configure appear in the BLF Presence Group relationship pane when you click Save. All groups that use system default permission field do not appear.
Before you begin

Configure BLF Presence Groups, on page 528

Procedure

Step 1
In the Cisco Unified CM Administration, choose Device > Phone, and click Add New. The Add a New Phone window appears.

Step 2
From the Phone Type drop-down list, select the type of phone that you want to associate BLF presence group to.

Step 3
Click Next.

Step 4
Configure the fields in the Phone Configuration window. See the online help for information about the fields and their configuration options.

Note
From the SUBSCRIBE Calling Search Space drop-down list, select a SUBSCRIBE calling search space to use for presence requests for the phone. All calling search spaces that you configure in Cisco Unified Communications Manager Administration appear 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 value of this field applies the default value as None. To configure a SUBSCRIBE calling search space specifically for this purpose, configure a calling search space as you configure all calling search spaces.

Step 5
Click Save.

What to do next

Associate BLF presence group with devices and users by performing the following subtasks:

- Associate BLF Presence Groups with SIP Trunk, on page 531
- Associate BLF Presence Groups with End User, on page 532
- Associate BLF Presence Groups with Application User, on page 533

Associate BLF Presence Groups with SIP Trunk

If digest authentication is not configured for the SIP trunk, you can configure the trunk to accept incoming subscriptions, but application-level authorization cannot be initiated, and Unified CM accepts all incoming requests before performing group authorization. When digest authentication is used with application-level authorization, Unified CM also authenticates the credentials of the application that is sending the BLF presence requests.

When there is a BLF presence request for a device that exists outside of the cluster, Cisco Unified Communications Manager queries the external device through the SIP trunk. If the watcher has permission to monitor the external device, the SIP trunk sends the BLF presence request to the external device, and returns BLF presence status to the watcher.
To use BLF presence group authorization with incoming presence requests on a SIP trunk, configure a presence group for the trunk, such as External_Presence_Serv_Group1, and configure the appropriate permissions to other groups inside the cluster.

If you configure both levels of authorization for SIP trunk presence requests, the BLF presence group for the SIP trunk gets used only when no BLF presence group is identified in the incoming request for the application.

**Before you begin**

Configure BLF Presence Groups, on page 528

**Procedure**

**Step 1**
In the Cisco Unified CM Administration, choose **Device > Trunk**, and click **Add New**. The **Trunk Configuration** window appears.

**Step 2**
From the **Trunk Type** drop-down list, select the type of phone that you want to associate BLF presence group. The value in the **Device Protocol** drop-down list populates automatically.

**Step 3**
Click **Next**.

**Step 4**
Configure the fields in the **Trunk Configuration** window. See the online help for information about the fields and their configuration options.

**Note**
To authorize the Unified CM system to accept incoming BLF presence requests from the SIP trunk, check the **Accept Presence Subscription** check box in the SIP Trunk Security Profile Configuration window. To block incoming presence requests on a SIP trunk, uncheck the check box. When you allow SIP trunk BLF presence requests, Unified CM accepts requests from the SIP user agent (SIP proxy server or external BLF presence server) that connects to the trunk. Consider digest authentication as optional when Unified CM is configured to accept BLF presence requests from a SIP trunk.

**Step 5**
Click **Save**.

**What to do next**

Associate BLF presence group with devices and users by performing the following subtasks:

- Associate BLF Presence Groups with Phone, on page 530
- Associate BLF Presence Groups with End User, on page 532
- Associate BLF Presence Groups with Application User, on page 533

**Associate BLF Presence Groups with End User**

An administrator associates BLF presence groups with end user for user directories and call lists and to configure extension mobility settings.
Before you begin
Configure BLF Presence Groups, on page 528

Procedure

Step 1 In the Cisco Unified CM Administration, choose User Management > End User, and click Add New. The End User Configuration window appears.
Step 2 Configure the fields in the End User Configuration window. See the online help for information about the fields and their configuration options.
Step 3 Click Save.

What to do next
Associate BLF presence group with devices and users by performing the following subtasks:

• Associate BLF Presence Groups with Phone, on page 530
• Associate BLF Presence Groups with SIP Trunk, on page 531
• Associate BLF Presence Groups with Application User, on page 533

Associate BLF Presence Groups with Application User
An administrator associates BLF Presence groups with an application user for external applications. These external applications send BLF presence requests that is SIP trunk or home on a proxy server which is connected on SIP trunk. For example, Web Dial, Meeting Place, conference servers, and presence servers.

Before you begin
Configure BLF Presence Groups, on page 528

Procedure

Step 1 In the Cisco Unified CM Administration, choose User Management > Application User, and click Add New. The Application User Configuration window appears.
Step 2 Configure the fields in the Application User Configuration window. See the online help for information about the fields and their configuration options.
Step 3 Click Save.

What to do next
Associate BLF presence group with devices and users by performing the following subtasks:

• Associate BLF Presence Groups with Phone, on page 530
• Associate BLF Presence Groups with SIP Trunk, on page 531
Accept BLF Presence Requests from External Trunks and Applications

To allow BLF presence requests from outside the cluster, configure the system to accept BLF presence requests from the external trunk or application. You can assign BLF presence groups to trunks and applications outside the cluster to invoke BLF presence group authorization.

Before you begin

Associate BLF presence group with devices and users by performing the following subtasks:

- Associate BLF Presence Groups with Phone, on page 530
- Associate BLF Presence Groups with SIP Trunk, on page 531
- Associate BLF Presence Groups with End User, on page 532
- Associate BLF Presence Groups with Application User, on page 533

Procedure

Step 1
In the Cisco Unified CM Administration, choose Device > Trunk, and click Add New. The Trunk Configuration window appears.

Step 2
To allow BLF presence requests from a SIP trunk, check the Accept Presence Subscription check box in the SIP Trunk Security Profile Configuration window.

Step 3
To enable application-level authorization for a SIP trunk application in addition to trunk-level authorization, check the following check boxes in the SIP Trunk Security Profile Configuration window:

- Enable Digest Authentication
- Enable Application Level Authorization

Note: You cannot check Enable Application Level Authorization unless Enable Digest Authentication is checked.

Step 4
Apply the profile to the trunk. Click Reset so that the changes to the trunk can take effect.

Note: If you checked Enable Application Level Authorization, check the Accept Presence Subscription check box in the Application User Configuration window for the application.

What to do next
Configure a Calling Search Space for Presence Requests, on page 535
Configure a Calling Search Space for Presence Requests

The SUBSCRIBE Calling Search space option allows you to apply a calling search space separate from the call-processing Calling Search Space for BLF presence requests. Select a different calling search space for presence requests, else the SUBSCRIBE Calling Search Space selects the None default option. The SUBSCRIBE Calling Search Space that is associated with an end user is used for extension mobility calls.

You apply the SUBSCRIBE Calling Search Space to the SIP trunk, phone, or end user. The SUBSCRIBE Calling Search Space that is associated with an end user is used for extension mobility calls.

Before you begin

Accept BLF Presence Requests from External Trunks and Applications, on page 534

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Calling Search Space.
Step 2 In the Calling Search Space configuration window, choose the calling search space from the SUBSCRIBE Calling Search Space drop-down list.
Step 3 Click Add New.
Step 4 In the Name field, enter a name.
Step 5 (Optional) In the Description field, enter a description to identify the calling search space.
Step 6 From the Available Partitions list, select one or multiple partitions, and click the arrow keys. The selected partitions appear in the Selected Partitions list.
Step 7 (Optional) To add or remove a partition from the Selected Partitions list, click the arrow keys next to the list box.
Step 8 Click Save.

All calling search spaces that you configure in Cisco Unified Communications Manager Administration appear in the SUBSCRIBE Calling Search Space drop-down list in the Trunk Configuration or Phone Configuration window.

What to do next

Configure a Phone Button Template for BLF and SpeedDial Buttons, on page 535

Configure a Phone Button Template for BLF and SpeedDial Buttons

You can configure BLF and SpeedDial buttons for a phone or user device profile. After you apply the template to the phone or device profile (and save the phone or device profile configuration), the Add a new BLF SD link appears in the Association Information pane in Cisco Unified Communications Administration.

Note

If the template does not support BLF and SpeedDials, the Add a new BLF SD link appears in the Unassigned Associated Items pane.
When an administrator decides to add or change a BLF and SpeedDial button for a SIP URI, the administrator ensures that the watcher is authorized to monitor that destination. If the system uses a SIP trunk to reach a SIP URI BLF target, the BLF presence group associated with the SIP trunk applies.

**Note**

You do not need to configure BLF presence groups or the Default Inter-Presence Group Subscription parameter for BLF and SpeedDials.

**Before you begin**

Configure a Calling Search Space for Presence Requests, on page 535

**Procedure**

**Step 1**
In the Cisco Unified CM Administration, choose Device > Device Settings > Phone Button Template.

**Step 2**
Click the Add New button. The Phone Button Template Configuration window appears.

**Step 3**
In the Button Template Name field, enter a name for the template.

**Step 4**
From the Phone Button Template drop-down list, select a template of phone button.

**Step 5**
Click Copy to create a new button template based on the layout of the selected button template.

**Step 6**
Click Save.

**What to do next**

Associate Button Template with a Device, on page 536

### Associate Button Template with a Device

You configure BLF and SpeedDial buttons for a phone or user device profile. The BLF value does not have to be on the cluster. For information on the Busy Lamp Field (BLF) status icons that display on the phone, see the Cisco Unified IP Phone documentation that supports your phone. To identify whether your phone supports BLF presence, see the Cisco Unified IP Phone documentation that supports your phone and this version of Cisco Unified Communications Manager.

**Before you begin**

Configure a Phone Button Template for BLF and SpeedDial Buttons, on page 535

**Procedure**

**Step 1**
In the Cisco Unified CM Administration, choose Device > Device Settings > Device Profile.

**Step 2**
Enter the search parameters to find the configured phone button templates, and click Find. The records matching all the search criteria appear.

**Step 3**
Click one of the records.
The Device Profile Configuration window appears.

**Step 4**
From the Phone Button Template list, select a configured phone button template.

**Step 5**
(Optional) Modify the values of the configured device.

**Step 6**
Click Save.

What to do next
Configure User Device Profile, on page 537

Configure User Device Profile

See the “BLF Presence with Extension Mobility” section of BLF Presence Interactions, on page 537 for details.

**Before you begin**
Associate Button Template with a Device, on page 536

**Procedure**

**Step 1**
In the Cisco Unified CM Administration, choose Device > Device Settings > Device Profile.

**Step 2**
Click Add New.
The Device Profile Configuration window appears.

**Step 3**
Configure the fields in Device Profile Configuration window. See the online help for information about the fields and their configuration options.

**Note**
If the phone button template that you applied to the phone or device profile does not support BLF and SpeedDials, the link does not appear in the Association Information pane, but appears in the Unassigned Associated Items pane.

**Step 4**
Click Save.

BLF Presence Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presence BLF with DNs on H.323 phones when the H.323 phone device serves as presence entity</td>
<td>When the H.323 phone is in the RING IN state, the BLF status gets reported as Busy. For the presence entities of phones that are running either SCCP or SIP and that are in the RING IN state, the BLF status gets reported as Idle.</td>
</tr>
</tbody>
</table>
Interaction Features

When the H.323 phone is not connected to Cisco Unified Communications Manager for any reason, such as the Ethernet cable is unplugged from the phone, the BLF status gets reported as Idle all the time. For presence entities of phones that are running either SCCP or SIP and that are not connected to Cisco Unified Communications Manager, the BLF status gets reported as Unknown.

Presence BLF with DNson H.323 phones when the H.323 phone device serves as presence entity

When you configure BLF and SpeedDial buttons in a user device profile in Cisco Unified Communications Manager Administration, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF and SpeedDial buttons after you log in to the device.

When the extension mobility user logs out, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF and SpeedDial buttons for the logout profile that is configured.

BLF Presence Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Presence</td>
<td>Cisco Unified Communications Manager Assistant does not support SIP presence.</td>
</tr>
<tr>
<td>BLF Presence Requests</td>
<td>Cisco Unified Communications Manager Administration rejects BLF presence requests to a directory number that is associated with a hunt pilot.</td>
</tr>
<tr>
<td>BLF on Call List Feature</td>
<td>The BLF on call list feature is not supported on the Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7960.</td>
</tr>
<tr>
<td>BLF and SpeedDials</td>
<td>The administrator ensures that the watcher is authorized to monitor the destination when configuring a BLF and SpeedDial. BLF presence group authorization does not apply to BLF and SpeedDials.</td>
</tr>
</tbody>
</table>

Note: BLF presence group authorization does not apply to any directory number or SIP URI that is configured as a BLF and Speed Dial that appears in a call list for phones that are running SIP.

If there is an overlapping DN, where there is the same extension in different partitions, the presence notifications are selected based on the order of the partitions configured within the SUBSCRIBE CSS assigned to the device.

For example, two BLF speed dials are configured on a phone.

- Extension 1234 in the "internal" partition
- Extension 1234 in the "external" partition

Whichever partition is listed first within the SUBSCRIBE CSS is the one that will provide BLF presence to the subscribed devices.
<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLF Presence Authorization</td>
<td>For Cisco Unified IP Phones with multiple lines, the phone uses the cached information that is associated with the line directory number for missed and placed calls to determine BLF presence authorization. If this call information is not present, the phone uses the primary line as the subscriber for BLF presence authorization. For BLF and SpeedDial buttons on Cisco Unified IP Phones with multiple lines, the phone uses the first available line as the subscriber.</td>
</tr>
<tr>
<td>Cisco Unified IP Phone</td>
<td>When a user monitors a directory number that is configured for Cisco Unified IP Phones 7960 and 7940 that are running SIP, the system displays a status icon for ‘not on the phone’ on the watcher device when the presence entity is off-hook (but not in a call connected state). These phones do not detect an off-hook status. For all other phone types, the system displays the status icon for ‘on the phone’ on the watcher device for an off-hook condition at the presence entity.</td>
</tr>
<tr>
<td>SIP Trunks</td>
<td>BLF presence requests and responses must route to SIP trunks or routes that are associated with SIP trunks. The system rejects BLF presence requests routing to MGCP and H323 trunk devices.</td>
</tr>
<tr>
<td>BLF Presence-supported Phones that are running SIP</td>
<td>For BLF presence-supported phones that are running SIP, you can configure directory numbers or SIP URIs as BLF and SpeedDial buttons. For BLF presence-supported phones that are running SCCP, you can only configure directory numbers as BLF and SpeedDial buttons.</td>
</tr>
<tr>
<td>Phones that are running SIP</td>
<td>For phones that are running SIP, BLF presence group authorization also does not apply to any directory number or SIP URI that is configured as a BLF and Speed Dial that appears in a call list.</td>
</tr>
</tbody>
</table>
Presence and Privacy Features

BLF Presence Restrictions
Call Display Restrictions Overview

Cisco Unified Communications Manager provides flexible configuration options that allow and also restrict the display of the number and name information for both calling and connected users. You can restrict connected numbers and names independently of each other.

You can configure connected number and name restrictions on the SIP trunk level or on a call-by-call basis. The SIP trunk level configuration overrides a call-by-call configuration.

For example, in a hotel environment, you may want to see the display information for calls that are made between a guest room and the front desk. However, for calls between guest rooms, you can restrict the call information to display on either phone.

Call Display Restrictions Configuration Task Flow

Before you begin

• Review Call Display Restrictions Interactions, on page 551

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify endpoints that support the Call Display Restrictions feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Partitions for Call Display Restrictions, on page 542</td>
<td>Configure partitions to create a logical grouping of directory numbers (DN) and route patterns with similar reachability characteristics. For example, in a hotel environment, you can a</td>
</tr>
</tbody>
</table>
Configure Partitions for Call Display Restrictions

Configure partitions to create a logical grouping of directory numbers (DNs) and route 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. You can configure multiple partition.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Class of Control &gt; Partition</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Add New</strong> to create a new partition.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the <strong>Partition Name, Description</strong> 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.</td>
</tr>
<tr>
<td>Step 4</td>
<td>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 (&quot;), percentage sign (%), ampersand (&amp;), backslash (), angle brackets (&lt;&gt;), or square brackets ([ ]). If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.</td>
</tr>
</tbody>
</table>

Optional. Configure Call Display Restrictions on SIP Trunks, on page 549
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**.

---

**What to do next**

Configure Calling Search Spaces for Call Display Restrictions, on page 543

### 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 46: 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 for Call Display Restrictions

Configure calling search spaces to identify the partitions that calling devices can search when they attempt to complete a call. Create calling search spaces for rooms, the front desk, other hotel extensions, the PSTN, and the room park range (for call park).
Before you begin

Configure Partitions for Call Display Restrictions, on page 542

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.
Step 8 Click Save.

What to do next

Configure the Service Parameter for Connected Number Display Restriction, on page 544

Configure the Service Parameter for Connected Number Display Restriction

The connected number display restriction restricts the connected line ID display to dialed digits only. This option addresses customer privacy issues as well as connected number displays that are meaningless to phone users.

Before you begin

Configure Calling Search Spaces for Call Display Restrictions, on page 543

Procedure

Step 1 In the Cisco Unified CM Administration, choose System > Service Parameters.
Step 2 Select the server where the Cisco CallManager service runs, and then select the Cisco CallManager service.
Step 3 Set the Always Display Original Dialed Number service parameter to True to enable this feature.
   The default value is False.
Step 4  (Optional) Set the **Name Display for Original Dialed Number When Translated** service parameter.

The default field shows the alerting name of the original dialed number before translation. You can change this parameter to show the alerting name of the dialed number after translation. This parameter is not applicable if the **Always Display Original Number** service parameter is set to **False**.

Step 5  Click **Save**.

---

**Configure Translation Patterns**

Cisco Unified Communications Manager uses translation patterns to manipulate dialed digits before it routes a call. In some cases, the system does not use the dialed number. In other cases, the public switched telephone network (PSTN) does not recognize the dialed number. For the Call Display Restrictions feature, calls are routed through different translation patterns before the calls are extended to the actual device.

**Before you begin**

*Configure the Service Parameter for Connected Number Display Restriction*, on page 544

**Procedure**

Step 1  In Cisco Unified CM Administration, choose **Call Routing > Translation Pattern**. The **Translation Pattern Configuration** window appears.

Step 2  Configure the fields in the **Translation Pattern Configuration** window. See the Related Topics section below for more information about the fields and their configuration options.

Step 3  Click **Save**.

---

**What to do next**

*Configure Phones for Call Display Restrictions*, on page 547

**Related Topics**

*Translation Pattern Fields for Call Display Restrictions*, on page 546
# Translation Pattern Fields for Call Display Restrictions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Translation Pattern</td>
<td>Enter the translation pattern, including numbers and wildcards. Do not use spaces. For example, for the 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>
<tr>
<td>Description</td>
<td>Enter a description for the translation pattern. The description can include up to 50 characters in any language, but it cannot include double quotes (&quot;), percentage sign (%), ampersand (&amp;), or angle brackets (&lt;&gt;).</td>
</tr>
<tr>
<td>Partition</td>
<td>From the drop-down list, choose the partition to associate with this translation pattern.</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>From the drop-down list, choose the calling search space to associate with this translation pattern.</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• Default—Choose this option if you do not want to change the presentation of the calling line ID.</td>
</tr>
<tr>
<td></td>
<td>• Allowed—Choose this option if you want to display the phone number of the calling party.</td>
</tr>
<tr>
<td></td>
<td>• Restricted—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling party phone number.</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td></td>
<td>• Default—Choose this option if you do not want to change the presentation of the calling name.</td>
</tr>
<tr>
<td></td>
<td>• Allowed—Choose this option if you want to display the name of the calling party.</td>
</tr>
<tr>
<td></td>
<td>• Restricted—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling name.</td>
</tr>
</tbody>
</table>
Configure Phones for Call Display Restrictions

Use this procedure to associate phones with the partitions and the calling search spaces used for call display restrictions.

Before you begin

Configure Translation Patterns, on page 545

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>In the Cisco Unified CM Administration, choose Device &gt; Phone.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>a) Perform one of the following tasks: b) To modify the fields for an existing phone, enter search criteria and choose a phone from the resulting list. The Phone Configuration window appears. c) To add a new phone, click Add New. The Add a New Phone window appears.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>From the Calling Search Space drop-down list, choose the calling search space that you want the system to use when it determines how to route a dialed number.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Check the Ignore presentation indicators (internal calls only) check box to ignore any presentation restriction on internal calls.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click Save. The phone is added to the database.</td>
</tr>
</tbody>
</table>
Step 6  To associate the added phone to a directory number, choose Device > Phone, enter search parameters to search the phone that you added.

Step 7  In the Find and List Phones window, click the phone name. The Phone Configuration window appears.

Step 8  From the Association pane, click the phone name to add or modify the directory number. The Directory Number Configuration window appears.

Step 9  In the Directory Number Configuration window, add or modify the value of directory number in the Directory Number text box, and select a value in the Route Partition drop-down list.

Step 10  Click Save.

---

**Phone Configuration Example**

Configure phone A (Room-1) with partition P_Room and device/line calling search space CSS_FromRoom

{ P_Phone, CSS_FromRoom} : 221/Room-1

Configure phone B (Room-2) with partition P_Room and device/line calling search space CSS_FromRoom

{ P_Phone, CSS_FromRoom} : 222/Room-2

Configure phone C (Front Desk-1) with partition P_FrontDesk and device/line calling search space CSS_FromFrontDesk and Ignore Presentation Indicators check box enabled

{ P_FrontDesk, CSS_FromFrontDesk, IgnorePresentationIndicators set} : 100/Reception

Configure phone D (Front Desk-2) with partition P_FrontDesk and device/line calling search space CSS_FromFrontDesk and Ignore Presentation Indicators check box enabled

{ P_FrontDesk, CSS_FromFrontDesk, IgnorePresentationIndicators set} : 200/Reception

Configure phone E (Club) with partition P_Club and calling search space CSS_FromClub

{ P_Club, CSS_FromClub} : 300/Club

---

**What to do next**

Configure the PSTN Gateway for Call Display Restrictions, on page 548

**Configure the PSTN Gateway for Call Display Restrictions**

Associate the PSTN gateway with the partitions and the calling search spaces that you want to use for call display restrictions.

**Before you begin**

Configure Phones for Call Display Restrictions, on page 547
**Procedure**

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

**Step 2**
Enter search criteria and choose the PSTN gateway from the resulting list. The **Gateway Configuration** window appears.

**Step 3**
From the **Calling Search Space** drop-down list, choose the calling search space that you want the system to use when it determines how to route an incoming call from the PSTN.

**Step 4**
Click **Save** and **Reset** to apply the configuration changes.

**Step 5**
(Optional) To associate the available trunk or gateway, in Cisco Unified Communications Manager Administration, choose **SIP Route Pattern**, and select a SIP trunk or route list from the **SIP Trunk/Route List** drop-down box.

**Gateway Configuration Example**
Configure PSTN Gateway E with route pattern P_PSTN and calling search space CSS_FromPSTN {CSS_FromPSTN}, RoutePattern {P_PSTN}

**What to do next**
(Optional) **Configure Call Display Restrictions on SIP Trunks, on page 549**

### Configure Call Display Restrictions on SIP Trunks

You can configure connected number and name restrictions on the SIP trunk level. The SIP trunk-level configuration overrides call-by-call configuration.

**Before you begin**
(Optional) **Configure the PSTN Gateway for Call Display Restrictions, on page 548**

**Procedure**

**Step 1**
In the Cisco Unified CM Administration, choose **Device > Trunk**. The **Find and List Trunks** window appears.

**Step 2**
Enter search criteria and click **Find**.

**Step 3**
Select the name of the trunk that you want to update.

**Step 4**
Configure the fields in the **SIP Trunk Configuration** window. See the Related Topics section below for more information about the fields and their configuration options.

**Step 5**
Click **Save**.

**Related Topics**

**SIP Trunk Fields for Call Display Restrictions**, on page 550
## SIP Trunk Fields for Call Display Restrictions

### Table 47: Inbound Calls

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| **Calling Line ID Presentation** | From the drop-down list, choose one of the following options:  
• **Default**—Choose this option if you do not want to change the presentation of the calling line ID.  
• **Allowed**—Choose this option if you want to display the phone number of the calling party.  
• **Restricted**—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling party phone number.                                                                                                                                                                                                                     |
| **Calling Name Presentation**  | From the drop-down list, choose one of the following options:  
• **Default**—Choose this option if you do not want to change the presentation of the calling name.  
• **Allowed**—Choose this option if you want to display the name of the calling party.  
• **Restricted**—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling name.                                                                                                                                                                                                                     |
| **Calling Search Space**       | From the drop-down list, choose the calling search space to associate with this translation pattern.                                                                                                                                                                                                                                                                                                                                                                           |

### Table 48: Outbound Calls

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| **Connected Line ID Presentation** | From the drop-down list, choose one of the following options:  
• **Default**—Choose this option if you do not want to change the presentation of the connected line ID.  
• **Allowed**—Choose this option if you want to display the phone number of the connected party.  
• **Restricted**—Choose this option if you want Cisco Unified Communications Manager to block the display of the connected party phone number.                                                                                                                                                                                                                     |
From the drop-down list, choose one of the following options:

- **Default**—Choose this option if you do not want to change the presentation of the connected name.
- **Allowed**—Choose this option if you want to display the name of the connected party.
- **Restricted**—Choose this option if you want Cisco Unified Communications Manager to block the display of the connected name.

### Call Display Restrictions Interactions

This section describes how the Call Display Restrictions feature interacts with Cisco Unified Communications Manager applications and call processing features.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
</table>
| Call Park     | When you use the Call Display Restrictions feature with Call Park, you must configure an associated translation pattern for each individual call park number to preserve the Call Display Restrictions feature. You cannot configure a single translation pattern to cover a range of call park numbers. Consider the following scenario as an example:  

1. The system administrator creates a call park range of 77x and places it in a partition called P_ParkRange. (The phones in the guest rooms can see that the P_ParkRange partition is made visible to the phones in the guest rooms by inclusion of it in the calling search space of the phones [CSS_FromRoom]).  

2. The administrator configures a separate translation pattern for each call park directory number and configures the display fields to Restricted. (In the current scenario, the administrator creates translations patterns for 770, 771, 772...779.)  

**Note** For the Call Display Restrictions feature to work correctly, the administrator must configure separate translation patterns and not a single translation pattern for a range of numbers (such as 77x or 77[0-9]).  

3. Room-1 calls Room-2.  

4. Room-2 answers the call, and Room-1 parks the call.  

5. When Room-1 retrieves the call, Room-2 does not see Room-1 call information display.  

See [Call Park Overview](#)
Interaction Feature

When you use Call Display Restrictions, you restrict the display information for the list of participants in a conference.

See Ad Hoc Conferencing Overview

Conference List

When you use Call Display Restrictions with features, such as conference and voice mail, the call information display on the phones reflects that status. For example, when the conference feature is invoked, the call information display shows To Conference. When voice mail is accessed by choosing the Messages button, the call information display shows To Voicemail.

Conference and Voice Mail

To use Call Display Restrictions with Extension Mobility, enable the Ignore Presentation Indicators (internal calls only) parameter in both the Cisco Unified Communications Manager Administration Phone Configuration window and the Cisco Unified Communications Manager Administration Device Profile Configuration window.

When you enable Call Display Restrictions with Extension Mobility, the presentation or restriction of the call information depends on the line profile that is associated with the user who is logged in to the device. The configuration that is entered in the user device profile (associated with the user) overrides the configuration that is entered in the phone configuration (of the phone that is enabled for Extension Mobility).

Extension Mobility

The Connected Number Display restriction applies to all calls that originate in the system. When this value is set to True, this field interacts with existing Cisco Unified Communications Manager applications, features, and call processing. This value applies to all calls that terminate inside or outside the system. The Connected Number Display is updated to show the modified number or redirected number when a call is routed to a Call Forward All or Call Forward Busy destination, or gets redirected through a call transfer or CTI application.

Call Forwarding

Translation Patterns—Duplicate entries are not allowed in translation patterns.
Do Not Disturb Overview

Do Not Disturb (DND) provides the following options:

- **Call Reject**—This option specifies that the incoming call gets rejected. 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.

- **Ringer Off**—This option turns off the ringer, but incoming call information gets presented to the device, so that the user can accept the call.

When DND is enabled, all new incoming calls with normal priority honor the DND settings for the device. High-priority calls, such as Cisco Emergency Responder (CER) calls, or calls with Multilevel Precedence and Preemption (MLPP), ring on the device. Also, when you enable DND, the Auto Answer feature gets disabled.

Users can activate Do Not Disturb on the phone in the following ways:

- Softkey
- Feature button
- Cisco Unified Communications Self-Care Portal

You can also enable or disable the feature on a per-phone basis from within Cisco Unified Communications Manager.

Phone Behavior

When you enable Do Not Disturb, the Cisco Unified IP Phone displays the message “Do Not Disturb is active”. Some Cisco Unified IP Phones display DND status icons. For details on how individual phone models use Do Not Disturb, consult the user guide for that particular phone model.
When you activate DND, you can still receive incoming call notifications on the phone as specified by the Incoming Call Alert settings in Cisco Unified Communications Manager Administration, but the phone will not ring, except for high-priority calls (such as Cisco Emergency Responder and MLPP calls). Also, if you enable DND while the phone is ringing, the phone stops ringing.

**Status Notifications**

Do Not Disturb is supported on both SIP and Cisco Skinny Call Control Protocol (SCCP) devices.

SIP phones use the SIP PUBLISH method to signal a DND status change to Cisco Unified Communications Manager. Cisco Unified Communications Manager uses a Remote-cc REFER request to signal a DND status change to the SIP phone.

SCCP phones use SCCP messaging to signal a DND status change to Cisco Unified Communications Manager.

### Do Not Disturb Configuration 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 1</td>
<td>Run a Phone Feature List report from Cisco Unified Reporting to determine which phones support Do Not Disturb.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Busy Lamp Field Status, on page 555</td>
<td>Configure the Busy Lamp Field status service parameter.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure Do Not Disturb on a Common Phone Profile, on page 555</td>
<td>Optional. Configure Do Not Disturb against a Common Phone Profile. The profile allows you to apply Do Not Disturb settings to a group of phones in your network.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Apply Do Not Disturb Settings to the Phone, on page 556.</td>
<td>Apply Do Not Disturb settings to the phone.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Depending on whether your phone uses softkeys or feature buttons, perform either of the following tasks:</td>
<td>Add a Do Not Disturb feature button or softkey to your phone.</td>
</tr>
<tr>
<td>- Configure a Do Not Disturb Feature Button, on page 557</td>
<td></td>
</tr>
<tr>
<td>- Configure a Do Not Disturb Softkey, on page 558</td>
<td></td>
</tr>
</tbody>
</table>
Configure Busy Lamp Field Status

Configure how the Busy Lamp Field (BLF) status depicts Do Not Disturb by setting the BLF Status Depicts DND service parameter. To set the BLF status, do the following:

Before you begin
Generate a Phone Feature List, on page 1

Procedure

Step 1 In Cisco Unified CM Administration, choose System > Service Parameters.
Step 2 Choose the Cisco CallManager service for the server that you want to configure.
Step 3 In the Clusterwide Parameters (System - Presence) pane, specify one of the following values for the BLF Status Depicts DND service parameter:
   - True—If Do Not Disturb is activated on the device, the BLF status indicator for the device or line appearance reflects the Do Not Disturb state.
   - False—If Do Not Disturb is activated on the device, the BLF status indicator for the device or line appearance reflects the actual device state.

What to do next
Perform one of the following procedures:
Configure Do Not Disturb on a Common Phone Profile, on page 555
Apply Do Not Disturb Settings to the Phone, on page 556

Configure Do Not Disturb on a Common Phone Profile

Common Phone Profiles allow you to configure Do Not Disturb settings and then apply those settings to a group of phones in your network that use that profile.

Before you begin
Configure Busy Lamp Field Status, on page 555

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Common Phone Profile.
Step 2 From the DND Option drop-down list box, choose how you want the Do Not Disturb feature to handle incoming calls.
   - Call Reject—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.
• **Ringer Off**—This option turns off the ringer, but incoming call information gets presented to the device, so the user can accept the call.

**Note**  For mobile phones and dual-mode phones, you can only choose the Call Reject option.

**Step 3**  From the In**coming Call Alert** drop-down list box, choose how you want to alert phone users of incoming calls while Do Not Disturb is turned on.

• **Disable**—Both beep and flash notification of a call are for disabled. If you configured the DND Ringer Off option, incoming call information still gets displayed. However, for the DND Call Reject option, no call alerts display, and no information gets sent to the device.

• **Flash Only**—The phone flashes for incoming calls.

• **Beep Only**—The phone displays a flash alert for incoming calls.

**Step 4**  Click **Save**.

---

**What to do next**

*Apply Do Not Disturb Settings to the Phone, on page 556*

---

## Apply Do Not Disturb Settings to the Phone

This procedure describes how to apply Do Not Disturb settings on your Cisco Unified IP Phones. You can apply DND settings through the **Phone Configuration** window in Cisco Unified CM Administration, or you can apply DND settings to a Common Phone Profile and then apply that profile to your phone.

**Before you begin**

If you are using a Common Phone Profile, complete **Configure Do Not Disturb on a Common Phone Profile, on page 555**.

Otherwise, complete **Configure Busy Lamp Field Status, on page 555**.

**Procedure**

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

**Step 2**  Click **Find** and select the phone on which you want to configure Do Not Disturb.

**Step 3**  If you want to apply Do Not Disturb settings from a Common Phone Profile, from the **Common Phone Profile** drop-down list box, choose the profile on which you have configured Do Not Disturb.

**Step 4**  Check the **Do Not Disturb** check box to enable Do Not Disturb on the phone.

**Step 5**  In the **DND Option** drop-down list box, specify from the following options how you want the DND feature to handle incoming calls.

• **Call Reject**—No incoming call information gets presented to the user. Depending on the configuration, the phone either plays a beep or displays a flash notification.

• **Ringer Off**—Incoming call information gets presented to the device so that the user can accept the call, but the ringer is turned off.

• **Use Common Profile Setting**—The Do Not Disturb setting for the Common Phone Profile that is specified for this device gets used.
For 7940/7960 phones that are running SCCP, you can only choose the Ringer Off option. For mobile devices and dual-mode phones, you can only choose the Call Reject option. When you activate DND Call Reject on a mobile device or dual-mode phone, no call information gets presented to the device.

**Note**

For 7940/7960 phones that are running SCCP, you can only choose the Ringer Off option. For mobile devices and dual-mode phones, you can only choose the Call Reject option. When you activate DND Call Reject on a mobile device or dual-mode phone, no call information gets presented to the device.

**Step 6**

In the **DND Incoming Call Alert** drop-down list box, specify from the following options how you want the phone to display an incoming call when DND is turned on.

- **None**—The DND Incoming Call Alert setting from the Common Phone Profile gets used for this device.
- **Disable**—For DND Ringer Off, both beep and flash notifications are disabled, but incoming call information is still displayed. For Call Reject, beep and flash notifications are disabled, and no incoming call information gets passed to the device.
- **Beep only**—For incoming calls, the phone plays a beep tone only.
- **Flash only**—For incoming calls, the phone displays a flash alert.

**Step 7**

Click **Save**.

---

**What to do next**

Complete either of the following procedures:

- Configure a Do Not Disturb Feature Button, on page 557
- Configure a Do Not Disturb Softkey, on page 558

**Configure a Do Not Disturb Feature Button**

Follow these steps to add a Do Not Disturb feature button to your Cisco Unified IP Phone.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Phone Button Template for Do Not Disturb, on page 557</td>
<td>Create a phone button template that includes the Do Not Disturb button.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate a Button Template with a Phone, on page 237</td>
<td>Associate the Do Not Disturb button template to a phone.</td>
</tr>
</tbody>
</table>

**Configure Phone Button Template for Do Not Disturb**

Follow this procedure to configure a phone button template that includes the Do Not Disturb button.

**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:
   - If you modified a template that is already associated with devices, click Apply Config to restart the devices.
   - If you created a new softkey template, associate the template with the devices and then restart them.

What to do next
Associate a Button Template with a Phone, on page 237

Associate Button Template with Phone

Before you begin
Configure Phone Button Template for Do Not Disturb, on page 557

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.

What to do next
(Optional) Configure a Do Not Disturb Softkey, on page 558 or

Configure a Do Not Disturb Softkey

Optional. If your phone uses softkeys, perform the tasks in the following task flow to add a Do Not Disturb softkey to the phone.
Configure Softkey Template for Do Not Disturb

Perform these steps to configure a softkey template that includes the Do Not Disturb softkey.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Softkey Template for Do Not Disturb, on page 559</td>
</tr>
</tbody>
</table>
| **Step 2** | Perform either of the following procedures:  
  - Associate a Softkey Template with a Common Device Configuration, on page 560  
  - Associate Softkey Template with a Phone, on page 561 | You can associate the softkey to a Common Device Configuration and then associate that configuration to a group of phones, or you can associate the softkey template directly to a phone. |

**Configure Softkey Template for Do Not Disturb**

Perform these steps to configure a softkey template that includes the Do Not Disturb softkey.

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:  
  - If you modified a template that is already associated with devices, click **Apply Config** 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 procedures to add the softkey template to a phone.

Associate a Softkey Template with a Common Device Configuration, on page 560
Associate Softkey Template with a Phone, on page 561

Associate a Softkey Template with a Common Device Configuration

When you associate the Do Not Disturb (DND) softkey template to a Common Device Configuration you can add the DND softkey to a group of Cisco Unified IP Phones that use that Common Device Configuration.

Before you begin
Configure Softkey Template for Do Not Disturb, on page 559

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Add Softkey Template to Common Device Configuration, on page 560</td>
<td>Associate the DND softkey template to a Common Device Configuration.</td>
</tr>
<tr>
<td>Step 2 Associate Common Device Configuration with Phone, on page 561</td>
<td>Add the DND softkey to a phone by associating the Common Device Configuration to the phone.</td>
</tr>
</tbody>
</table>

Add Softkey Template to 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.

---

**What to do next**

*Associate Common Device Configuration with Phone, on page 561*

---

### Associate Common Device Configuration with Phone

**Before you begin**

*Associate a Softkey Template with a Common Device Configuration, on page 560*

**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>Device &gt; Phone</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> and select the phone device to add the softkey template.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Common Device Configuration</strong> drop-down list, choose the common device configuration that contains the new softkey template.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Reset</strong> to update the phone settings.</td>
</tr>
</tbody>
</table>

---

### Associate Softkey Template with a Phone

Perform this procedure if you have configured a softkey template with the Do Not Disturb softkey and you want to associate that softkey template to a phone.

**Before you begin**

*Configure Softkey Template for Do Not Disturb, on page 559*

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Phone</strong>.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> to select the phone to add the softkey template.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Softkey Template</strong> drop-down list, choose the template that contains the new softkey.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Press <strong>Reset</strong> to update the phone settings.</td>
</tr>
</tbody>
</table>
Do Not Disturb Interactions and Restrictions

This section provides information about Do Not Disturb interactions and restrictions.

Interactions

The following table describes feature interactions with the Do Not Disturb (DND) feature. Unless otherwise stated, the interactions apply to both the DND Ringer Off and the DND Call Reject option.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction with Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Forward All</td>
<td>On Cisco Unified IP Phones, the message that indicates that the Do Not Disturb (DND) feature is active takes priority over the message that indicates that the user has new voice messages. However, the message that indicates that the Call Forward All feature is active has a higher priority than DND.</td>
</tr>
</tbody>
</table>
| Park Reversion        | For locally parked calls, Park Reversion overrides DND. If Phone A has DND turned on, and a call is parked, the park reversion to Phone A occurs and Phone A rings. For remotely parked calls, DND overrides Park Reversion:  
  • If Phone A activates DND Ringer Off and shares a line with Phone A-prime, when Phone A-prime parks the call, park reversion on Phone A honors the DND settings and does not ring.  
  • If Phone A activated DND Call Reject, the park reversion is not presented to Phone A. |
| Pickup                | For locally placed Pickup requests, Pickup overrides DND. If Phone A has DND turned on, and has initiated any type of Pickup, the Pickup call presents normally, and Phone A rings. For remotely placed Pickup requests, DND overrides Pickup as follows:  
  • If Phone A is in DND Ringer Off mode and shares a line with Phone A-prime, when Phone A-prime initiates Pickup, the Pickup call to Phone A honors the DND settings and Phone A does not ring.  
  • If Phone A is in DND Call Reject mode, the Pickup call is not presented to Phone A. |
| Hold Reversion and Intercom | Hold Reversion and Intercom override DND, and the call gets presented normally. |
| MLPP and CER          | Multilevel Precedence and Preemption (phones that are running SCCP) and Cisco Emergency Responder calls override DND. Multilevel Precedence and Preemption and Cisco Emergency Responder calls get presented normally, and the phone rings. |
### Interaction with Do Not Disturb

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction with Do Not Disturb</th>
</tr>
</thead>
</table>
| Call Back     | For the originating side, callback overrides DND. When the activating device is on DND mode, the callback notification (both audio and visual) is still presented to the user. For the terminating side, DND overrides callback as follows:  
  - When the terminating side is on DND Ringer Off, the Callback Available screen is sent after the terminating side goes off hook and on hook.  
  - When the terminating side is on DND Call Reject, and is available, a new screen is sent to the activating device as “<DirectoryNumber> has become available but is on DND-R” if the activating device is in same cluster. Callback available notification is sent only after the terminating side disables DND Call Reject. |
| Pickup Notification | For the DND Ringer Off option, only visual notification gets presented to the device.  
For the DND Call Reject option, no notification gets presented to the device. |
| Hunt List     | If a device in a Hunt List has DND Ringer Off activated, the call is still presented to the user. However, the DND Incoming Call Alert settings would still apply. If a device in a Hunt List has DND Call Reject activated, any calls to that Hunt List will go to the next member and will not get sent to this device. |
| Extension Mobility | For Extension Mobility, the device profile settings include DND incoming call alert and DND status. When a user logs in and enables DND, the DND incoming call alert and DND status settings get saved, and these settings get used when the user logs in again.  
**Note** When a user who is logged in to Extension Mobility modifies the DND incoming call alert or DND status settings, this action does not affect the actual device settings. |

### Restrictions

Some restrictions apply to DND usage, depending on the phone or device type in use.

- The following phone models and devices that are running SCCP support only the DND Ringer Off option:
  - Cisco Unified IP Phone 7940
  - Cisco Unified IP Phone 7960
  - Cisco IP Communicator

**Note** Cisco Unified IP Phones 7940 and 7960 that run SIP use their own implementation of Do Not Disturb, which is backward compatible.

- The following phone models and devices support only the DND Call Reject option:
• Mobile devices (dual mode)
• Remote Destination Profile
• Cisco Unified Mobile Communicator

Do Not Disturb Troubleshooting

This section provides troubleshooting information for Cisco Unified IP Phones (SCCP and SIP).

For SIP phones, use the following information for troubleshooting:

• debugs: sip-dnd, sip-messages, dnd-settings
• show: config, dnd-settings
• sniffer traces

For SCCP phones, use the following information for troubleshooting:

• debug: jvm all info
• sniffer traces

Troubleshooting Errors

The following table describes how to troubleshoot errors with Do No Disturb.

<table>
<thead>
<tr>
<th>Symptom</th>
<th>Actions</th>
</tr>
</thead>
</table>
| DND softkey does not display or DND feature button does not display | • Verify that the softkey or button template for this phone includes DND.  
  • Capture a sniffer trace and verify that the phone gets the correct softkey or button template.  
  • Verify that the phone firmware is Version 8.3(1) or later. |
| BLF speed dial does not show DND status       | • Verify that the BLF DND is set to enabled in Enterprise parameters.  
  • Capture a sniffer trace and verify that the phone gets the correct notification message.  
  • Verify that the phone firmware is Version 8.3(1) or later. |
Privacy Overview

The Privacy feature allows you to enable or disable the capability of users with phones that share the same line (DN) to view call status and to barge into the call. You can enable or disable privacy for each phone or for all phones. By default, the system enables privacy for all phones in the cluster.

When the device that is configured for privacy registers with Cisco Unified Communications Manager, the feature button on the phone that is configured with privacy gets labeled, and the status is indicated through an icon. If the button has a lamp, it comes on.

When the phone receives an incoming call, the user makes the call private (so the call information does not display on the shared line) by pressing the Privacy feature button. The Privacy feature button toggles between On and Off.

To verify if your Cisco Unified IP Phone supports Privacy, see the user documentation for your phone model.

Privacy on Hold

Privacy on Hold allows you to enable or disable the capability of users with phones that share the same line (DN) to view call status and retrieve calls on hold.

You can enable or disable Privacy on Hold for specific phones or all the phones. Privacy on Hold activates automatically on all private calls when Privacy on Hold is enabled. By default, the system disables Privacy on Hold for all phones in the cluster.

To activate Privacy on Hold, users press the Hold softkey or Hold button while on a private call. To return to the call, users press the Resume softkey. The phone that puts the call on hold displays the status indicator for a held call; shared lines display the status indicators for a private and held call.
Privacy Configuration Task Flow

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that support the Privacy feature.</td>
</tr>
<tr>
<td>Step 2 Enable Privacy Cluster-wide, on page 566</td>
<td>Enable Privacy by default for all the phones in the cluster.</td>
</tr>
<tr>
<td>Step 3 Enable Privacy for a Device, on page 566</td>
<td>Enable Privacy for specific devices.</td>
</tr>
<tr>
<td>Step 4 Configure Privacy Phone Button Template, on page 567</td>
<td>Configure Privacy phone button template for a device.</td>
</tr>
<tr>
<td>Step 5 Associate Privacy Phone Button Template with a Phone, on page 568</td>
<td>Associate the phone button template with a user.</td>
</tr>
<tr>
<td>Step 6 Configure Shared Line Appearance, on page 568</td>
<td>Configure the shared line appearance.</td>
</tr>
<tr>
<td>Step 7 (Optional) Configure Privacy on Hold, on page 569</td>
<td>Configure Privacy on Hold.</td>
</tr>
</tbody>
</table>

Enable Privacy Cluster-wide

Perform these steps to enable Privacy by default for the entire cluster.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Service Parameters. The Service Parameter Configuration window appears.
Step 2 From the Server drop-down list, choose the server that is running the Cisco CallManager service.
Step 3 From the Service drop-down list, choose Cisco CallManager.
Step 4 From the Privacy Setting drop-down list, choose True.
Step 5 Click Save.

Enable Privacy for a Device

Before you begin

Ensure that the phone model supports Privacy. For more information, see Generate a Phone Feature List, on page 1.
Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Specify search criteria and click Find. The phone search results appear.
Step 3  Select the phone.
Step 4  From the Privacy drop-down list, select Default.
Step 5  Click Save.

What to do next
Configure Privacy Phone Button Template, on page 567

Configure Privacy Phone Button Template

Before you begin
Enable Privacy for a Device, on page 566

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:
   • If you modified a template that is already associated with devices, click Apply Config to restart the devices.
   • If you created a new softkey template, associate the template with the devices and then restart them.

What to do next
Associate Privacy Phone Button Template with a Phone, on page 568
### Associate Privacy Phone Button Template with a Phone

**Before you begin**

Configure Privacy Phone Button Template, on page 567

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Find** to display the list of configured phones.
3. Choose the phone to which you want to add the phone button template.
4. In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.
5. Click **Save**.

A dialog box is displayed with a message to press **Reset** to update the phone settings.

**What to do next**

Configure Shared Line Appearance, on page 568

### Configure Shared Line Appearance

**Before you begin**

Associate Privacy Phone Button Template with a Phone, on page 568

**Procedure**

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

The **Find and List Phones** window appears.

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

A list of phones that match the search criteria is displayed.

3. Choose the phone for which you want to configure shared line appearance.

The **Phone Configuration** window is displayed.

4. Click **Add a new DN link** in the Association Information area on the left side of the **Phone Configuration** window.

The **Directory Number Configuration** window appears.

5. Enter the **Directory Number** and choose the **Route Partition** to which the directory number belongs.

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

7. Repeat Step 3, on page 568 to Step 6, on page 568 for all the phones for which you want to create a shared line appearance.
### Configure Privacy on Hold

**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>System &gt; Service Parameters</strong>. The Service Parameter Configuration window appears.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>From the <strong>Server</strong> drop-down list, choose the server that is running the Cisco CallManager service.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>From the <strong>Service</strong> drop-down list, choose <strong>Cisco CallManager</strong>.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Set the <strong>Enforce Privacy Setting on Held Calls</strong> service parameter to <strong>True</strong>.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

### Privacy Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| CTI         | • CTI does not support Privacy through APIs that TAPI and JTAPI applications invoke. CTI generates events when Privacy is enabled or disabled from an IP phone by using the Privacy feature button.  
  • CTI does not support Privacy on Hold through APIs that TAPI/JTAPI applications invoke. CTI generates events when a Privacy-enabled call is put on hold and when Privacy gets enabled or disabled on held calls from an IP phone by using the Privacy feature button. |
Private Line Automatic Ringdown

- Private Line Automatic Ringdown Overview, on page 571
- Private Line Automatic Ringdown Configuration Task Flow for SCCP Phones, on page 571
- Private Line Automatic Ringdown Configuration Task Flow for SIP Phones, on page 574
- Private Line Automatic Ringdown Troubleshooting, on page 575

Private Line Automatic Ringdown Overview

The Private Line Automatic Ringdown (PLAR) feature configures a phone so that when the user goes off hook (or the NewCall softkey or line key gets pressed), the phone immediately dials a preconfigured number. The phone user cannot dial any other number from the phone line that gets configured for PLAR.

PLAR works with features such as Barge, cBarge, or single button Barge. If you use PLAR with a feature, you must configure the feature as described in the feature documentation, and you must configure the PLAR destination, which is a directory number that is used specifically for PLAR.

Private Line Automatic Ringdown Configuration Task Flow for SCCP Phones

Perform the following tasks to configure Private Line Automatic Ringdown (PLAR) on SCCP phones.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Create Partition, on page 572</td>
<td>Create a partition for the PLAR destination. The only directory number that you can assign to this partition is the PLAR destination.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign Partitions to Calling Search Spaces, on page 572</td>
<td>Assign the partition to a unique CSS, and a CSS that includes the PLAR destination device.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Assign Partition to the Private Line Automatic Ringdown Destination, on page 573</td>
<td>Assign the null partition and a CSS to your PLAR destination directory number.</td>
</tr>
</tbody>
</table>
Create Partition

Create a new partition for the Private Line Automatic Ringdown (PLAR) destination. For the feature to work, only the null translation pattern that you configure for PLAR can be assigned to this partition.

Procedure

Step 1
In Cisco Unified CM Administration, choose **Call Routing > Class of Control > Partition**.

Step 2
Click **Add New**.

Step 3
In the **Name** field, enter a partition name and a description separated by a comma.

Step 4
Click **Save**.

What to do next

Assign Partitions to Calling Search Spaces, on page 572

Assign Partitions to Calling Search Spaces

For Private Line Automatic Ringdown (PLAR) on SCCP phones, you must configure two calling search spaces (CSS):

- The first CSS should include the new partition for the null translation pattern as well as a partition that routes to the destination phone.
- The second CSS should include only the new partition for the null translation pattern.

Before you begin

Create Partition, on page 572

Procedure

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

Step 2
Click **Find** and select the calling search space for the PLAR destination device.

Step 3
Use the arrows to move both of the following partitions to the **Selected Partitions** list box: the new partition that you created for the null translation pattern and a partition that routes to the destination device.

Step 4
Click **Save**.

Step 5
Click **Add New**.

Step 6
Enter a name and description for the calling search space.
Step 7  Use the arrows to move the new partition to the **Selected Partitions** list box.
Step 8  Click **Save**.

---

**What to do next**

Assign Partition to the Private Line Automatic Ringdown Destination, on page 573

---

## Assign Partition to the Private Line Automatic Ringdown Destination

When configuring Private Line Automatic Ringdown (PLAR) on SCCP phones, assign a null partition to the directory number that you want to use as the PLAR destination.

---

**Note**

Each PLAR destination directory number must have its own unique partition. Do not add any other directory numbers to the null partition that you created for the PLAR destination.

---

**Before you begin**

Assign Partitions to Calling Search Spaces, on page 572

---

**Procedure**

1. In Cisco Unified CM Administration, choose **Call Routing > Directory Number**.
2. Click **Find** and select the directory number that you want to use as the PLAR destination.
3. In the **Route Partition** field, select a partition that you created for your PLAR destination.
4. In the **Calling Search Space** drop-down list box, select the CSS that includes both the null partition and the destination device.
5. Click **Save**.

---

**What to do next**

Configure Translation Pattern for Private Line Automatic Ringdown on Phones, on page 573

---

## Configure Translation Pattern for Private Line Automatic Ringdown on Phones

To configure Private Line Automatic Ringdown (PLAR) on phones, configure a null translation pattern and assign the PLAR destination number to that translation pattern.

---

**Before you begin**

Assign Partition to the Private Line Automatic Ringdown Destination, on page 573
Private Line Automatic Ringdown Configuration Task Flow for SIP Phones

Perform these tasks to configure Private Line Automatic Ringdown (PLAR) on SIP Phones.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Create SIP Dial Rule for Private Line Automatic Ringdown, on page 574</td>
<td>Create a SIP dial rule for PLAR.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Assign Private Line Automatic Ringdown Dial Rule to SIP Phone, on page 575</td>
<td>Assign the PLAR dial rule to the phone.</td>
</tr>
</tbody>
</table>

Create SIP Dial Rule for Private Line Automatic Ringdown

To configure Private Line Automatic Ringdown (PLAR) on SIP phones, you must configure a SIP dial rule for your PLAR destination number.

**Before you begin**

Create Partition, on page 572
Assign Partitions to Calling Search Spaces, on page 572
Assign Partition to the Private Line Automatic Ringdown Destination, on page 573
Configure Translation Pattern for Private Line Automatic Ringdown on Phones, on page 573

**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>In Cisco Unified CM Administration, choose Call Routing &gt; Class of Control &gt; SIP Dial Rules.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click Add New.</td>
<td></td>
</tr>
</tbody>
</table>
Assign Private Line Automatic Ringdown Dial Rule to SIP Phone

You can configure Private Line Automatic Ringdown (PLAR) on SIP phones by assigning a PLAR-enabled SIP Dial Rule to the phone.

Before you begin
Create SIP Dial Rule for Private Line Automatic Ringdown, on page 574

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Find and select the phone on which you want to configure PLAR.
Step 3 From the SIP Dial Rules drop-down list box, choose the dial rule that you created for PLAR.
Step 4 Click Save.

Private Line Automatic Ringdown Troubleshooting

Troubleshooting Private Line Automatic Ringdown on SCCP Phones

<table>
<thead>
<tr>
<th>Symptom</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>The phone goes off hook and the user hears a fast busy (reorder) tone.</td>
<td>Make sure that the CSS that is assigned to the PLAR translation pattern contains the partition of the PLAR destination.</td>
</tr>
<tr>
<td>The phone goes off hook and receives dial tone.</td>
<td>Make sure that the CSS that is assigned to the phone contains the partition of the null PLAR translation pattern.</td>
</tr>
</tbody>
</table>
### Troubleshooting Private Line Automatic Ringdown on SIP Phones

<table>
<thead>
<tr>
<th>Symptom</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>The phone goes off hook and the user hears fast busy (reorder) tone.</td>
<td>Make sure that the CSS of the SIP phone can reach the PLAR destination.</td>
</tr>
<tr>
<td>The phone goes off hook and receives a dial tone.</td>
<td>Make sure that the SIP Dial Rule has been created and is assigned to the phone.</td>
</tr>
</tbody>
</table>
Secure Tone

Secure Tone Overview

The Secure Tone feature can configure a phone to play a secure indication tone when a call is encrypted. The tone indicates that the call is protected and that confidential information may be exchanged. The 2-second tone comprises three long beeps. If the call is protected, the tone begins to play on a protected phone as soon as the called party answers.

When the call is not protected, the system plays a nonsecure indication tone, which comprises six short beeps, on a protected phone.

Note

Only callers on protected phones can hear secure and nonsecure indication tones. Callers on phones that are not protected cannot hear these tones.

The secure and nonsecure indication tones are supported on the following types of calls:

- Intracluster to IP-to-IP calls
- Intercluster protected calls
- IP-to-Time-Division-Multiplexing (TDM) calls through a protected MGCP E1 PRI gateway

For video calls, the system plays secure and nonsecure indication tones on protected devices.

Note

For video calls, the user may first hear secure indication tone for the audio portion of the call and then nonsecure indication tone for overall nonsecure media.

A lock icon that is displayed on a Cisco Unified IP Phone indicates that the media are encrypted, but does not indicate that the phone has been configured as a protected device. However, the lock icon must be present for a protected call to occur.
Protected Device Gateways

You can configure only supported Cisco Unified IP Phones and MGCP E1 PRI gateways as protected devices in Cisco Unified Communications Manager.

Cisco Unified Communications Manager can also direct an MGCP Cisco IOS gateway to play secure and nonsecure indication tones when the system determines the protected status of a call.

Protected devices provide these functions:

- You can configure phones that are running SCCP or SIP as protected devices.
- Protected devices can call nonprotected devices that are either encrypted or nonencrypted. In such cases, the call specifies nonprotected and the system plays nonsecure indication tone to the phones on the call.
- When a protected phone calls another protected phone, but the media is not encrypted, the system plays a nonsecure indication tone to the phones on the call.

Secure Tone Prerequisites

- You must configure the MGCP gateway for SRTP encryption. Configure the gateway with this command: `mgcp package-capability srtp-package`.
- The MGCP gateway must specify an Advanced IP Services or Advanced Enterprise Services image (for example, c3745-adventerprisek9-mz.124-6.T.bin).

Secure Tone Configuration Task Flow

**Before you begin**

- Review Secure Tone Prerequisites, on page 578

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Generate a Phone Feature List, on page 1</td>
<td>Generate a report to identify devices that support the Secure Tone feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Phone As a Protected Device, on page 579</td>
<td>Configure the phone as a protected device.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Directory Number for Secure Tones, on page 579</td>
<td>Configure multiple calls and call waiting settings for the protected device.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Secure Tone Service Parameters, on page 580</td>
<td>Configure service parameters.</td>
</tr>
<tr>
<td>Step 5</td>
<td>(Optional) Configure MGCP E1 PRI Gateway, on page 580</td>
<td>This configuration allows the system to pass protected status of the call between Cisco Unified IP Phone endpoints and the protected PBX phones that connect to the MGCP gateway.</td>
</tr>
</tbody>
</table>
Configure Phone As a Protected Device

Before you begin
Generate a Phone Feature List, on page 1

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window is displayed.
Step 2 Click the phone for which you want to set secure tone parameters. The Phone Configuration window is displayed.
Step 3 From the Softkey Template drop-down list in the Device Information portion of the window, choose Standard Protected Phone.
   Note You must use a new sofkey template without supplementary service softkeys for a protected phone.
Step 4 Set the Join Across Lines option to Off.
Step 5 Check the Protected Device check box.
Step 6 From the Device Security Profile drop-down list (in the Protocol Specific Information portion of the window), choose a secure phone profile that is already configured in the Phone Security Profile Configuration window (System > Security Profile > Phone Security Profile).
Step 7 Click Save.

What to do next
Perform one of the following procedures:
   • Configure Directory Number for Secure Tones, on page 579
   • Configure MGCP E1 PRI Gateway, on page 580

Configure Directory Number for Secure Tones

Before you begin
Configure Phone As a Protected Device, on page 579

Procedure

Step 1 Locate the Association section on the Phone Configuration window.
Step 2 Select Add a new DN. The Directory Number Configuration window is displayed.
Step 3 Specify a directory number in the Directory Number field.
Step 4 In the Multiple Call/Call Waiting Settings on Device [device name] area of the Directory Number Configuration window, set the Maximum Number of Calls and Busy Trigger options to 1.
Configure Secure Tone Service Parameters

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose System > Service Parameters.
Step 2 From the Server drop-down list box, choose a server.
Step 3 From the Service drop-down list box, choose Cisco CallManager.
Step 4 In the Clusterwide Parameters (Feature - Secure Tone) area, set the Play Tone to Indicate Secure/Non-Secure Call Status option to True.
Step 5 Click Save.

Configure MGCP E1 PRI Gateway

If you want the system to pass the protected status of the call between Cisco Unified IP Phone endpoints and the protected PBX phones that connect to the MGCP gateway, follow these steps:

Before you begin
Configure Phone As a Protected Device, on page 579

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose Device > Gateway.
Step 2 Specify the appropriate search criteria and click Find.
Step 3 Choose a MGCP gateway.
The Gateway Configuration window appears.
Step 4 Set Global ISDN Switch Type to Euro.
Step 5 Configure the fields in the Gateway Configuration window. See the online help for more information about the fields and their configuration options.
Step 6 Click Save.
Step 7 Click the Endpoint icon that appears to the right of subunit 0 in the window. The Enable Protected Facility IE check box appears. Check this check box.
Secure Tone Interactions and Restrictions

Secure Tone Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Transfer, Conference, and Call Waiting</td>
<td>When the user invokes these features on a protected phone, the system plays a secure or nonsecure indication tone to indicate the updated status of the call.</td>
</tr>
<tr>
<td>Hold/Resume and Call Forward All</td>
<td>These features are supported on protected calls.</td>
</tr>
</tbody>
</table>

Secure Tone Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Extension Mobility and Join Across Line services</td>
<td>Cisco Extension Mobility and Join Across Line services are disabled on protected phones.</td>
</tr>
<tr>
<td>Shared-line configuration</td>
<td>Shared-line configuration is not available on protected phones.</td>
</tr>
<tr>
<td>Non-encrypted media</td>
<td>If the media between the Cisco Unified IP Phone and the MGCP E1 PRI gateway are not encrypted, the call drops.</td>
</tr>
</tbody>
</table>
PART XI

Custom Features

• Branding Customizations, on page 585
• Client Matter Codes and Forced Authorization Codes, on page 593
• Custom Phone Rings and Backgrounds, on page 599
• Music On Hold, on page 609
• Self Care Portal, on page 627
• Emergency Call Handler, on page 631
• Enterprise Groups, on page 645
Branding Customizations

- Branding Overview, on page 585
- Branding Prerequisites, on page 585
- Branding Task Flow, on page 586
- Branding File Requirements, on page 588

Branding Overview

The Branding feature lets you upload customized branding for Cisco Unified Communications Manager. Branding gets applied to the Cisco Unified CM Administration login and configuration windows. Among the items that you can modify include:

- Company logos
- Background colors
- Border colors
- Font colors

Append Logo in Self Care Portal

The Branding feature allows you to append your company logo to the Unified Communications Self Care Portal login page and to the user interface header. You must include the `branding_logo.png` file in your `branding.zip` file and upload the zip file into Cisco Unified Communications Manager. The logo displays in the Self Care Portal after you enable branding in Cisco Unified Communications Manager.

There is no option to customize background colors or fonts for the Self-Care portal.

Branding Prerequisites

You must create your `branding.zip` file that contains the specified folder structure and files. For details, see Branding File Requirements, on page 588.
Branding Task Flow

Complete these tasks to apply branding in Cisco Unified Communications Manager and the Unified Communications Self-Care Portal.

Before you begin

- Review Branding Prerequisites, on page 585

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure your branding settings using one of these procedures:</td>
<td>Apply branding across the Cisco Unified Communications Manager cluster.</td>
</tr>
<tr>
<td>- Enable Branding, on page 586</td>
<td></td>
</tr>
<tr>
<td>- Disable Branding, on page 587</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> Restart the Tomcat Service, on page 587</td>
<td>You must restart the Cisco Tomcat service for the new branding setting to get picked up by the Unified Communications Self-Care Portal.</td>
</tr>
</tbody>
</table>

Enable Branding

Use this procedure to enable branding customization for Cisco Unified Communications Manager. Branding updates appear even if the system is enabled for SAML Single Sign-On.

To enable branding, you must use the master administrator account with privilege level 4 access. This is the main administrator account that is created during installation.

Before you begin

Prepare your branding.zip file and save it in a location that Cisco Unified Communications Manager can access.

Procedure

1. Log in to Cisco Unified OS Administration.
2. Choose Software Upgrades > Branding.
3. Browse to your remote server and select the branding.zip file.
4. Click Upload File.
5. Click Enable Branding.

Note You can also enable branding by running the `utils branding enable` CLI command.
Step 6  Refresh your browser.
Step 7  Repeat this procedure on all Cisco Unified Communications Manager cluster nodes.

What to do next
If you want to append your company logo to the Self-Care Portal user interface:
  • Restart the Tomcat Service, on page 587

Disable Branding
Use this procedure to disable branding in your Cisco Unified Communications Manager cluster. You also need to disable branding if you want to remove your company logo from the Self-Care Portal.

Note
To disable branding, you must use the master administrator account with privilege level 4 access. This is the main administrator account that is created during installation.

Procedure

Step 1  Log in to Cisco Unified OS Administration.
Step 2  Choose Software Upgrades > Branding.
Step 3  Click Disable Branding.
  Note  You can also disable branding by running the utils branding disable CLI command.
Step 4  Refresh your browser.
Step 5  Repeat this procedure on all Cisco Unified Communications Manager cluster nodes.

What to do next
If you want to remove your company logo from the Self-Care Portal user interface:
  • Restart the Tomcat Service, on page 587

Restart the Tomcat Service
You must restart the Cisco Tomcat service for the branding updates to reflect in the Self-Care Portal.

Before you begin
Make sure that you have completed the following:
• To append your logo to the Self-Care Portal, you must first enable branding in Cisco Unified Communications Manager. The branding.zip upload file must include a 44x25 pixel branding_logo.png file with your company logo. For details, Enable Branding, on page 586.

• To remove your logo from the Self-Care Portal, you must disable branding in Cisco Unified Communications Manager. For details, Disable Branding, on page 587.

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Log in to the Command Line Interface.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Run the <code>utils service restart Cisco Tomcat</code> CLI command.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Repeat this procedure on all Cisco Unified Communications Manager cluster nodes.</td>
</tr>
</tbody>
</table>

What to do next

After the service restarts, refresh your browser to see the changes in the Self-Care Portal.

Branding File Requirements

Before you apply customized branding to your system, create your branding.zip file according to the prescribed specifications. On a remote server, create a Branding folder and fill the folder with the specified contents. Once you have added all the image files and subfolders, zip the entire folder and save the file as branding.zip.

There are two options for the folder structure, depending on whether you want to use a single image for the header or a combination of six images in order to create a graded effect for the header.

Table 49: Folder Structure Options

<table>
<thead>
<tr>
<th>Branding Option</th>
<th>Folder Structure</th>
</tr>
</thead>
<tbody>
<tr>
<td>Single Header Option</td>
<td>If you want a single image for the header background (callout item 3), your</td>
</tr>
<tr>
<td></td>
<td>branding folder must contain the following subfolders and image files:</td>
</tr>
<tr>
<td></td>
<td>Branding (folder)</td>
</tr>
<tr>
<td></td>
<td>ccmadmin (folder)</td>
</tr>
<tr>
<td></td>
<td>BrandingProperties.properties (properties file)</td>
</tr>
<tr>
<td></td>
<td>brandingHeader.gif (2048*1 pixel image)</td>
</tr>
<tr>
<td></td>
<td>ciscoLogo12pxMargin.gif (44*44 pixel image)</td>
</tr>
<tr>
<td></td>
<td>branding_logo.png (44*25 pixel image)</td>
</tr>
</tbody>
</table>
### Branding Option

**Graded Header Option**

If you want to create a graded image for the header background, you need six separate image files to create the graded effect. Your branding folder must contain these subfolders and files:

- **Branding(folder)**
  - **ccmadmin (folder)**
    - **BrandingProperties.properties (file)**
    - **brandingHeaderBegLTR.gif (652*1 pixel image)**
    - **brandingHeaderBegRTR.gif (652*1 pixel image)**
    - **brandingHeaderEndLTR.gif (652*1 pixel image)**
    - **brandingHeaderEndRTR.gif (652*1 pixel image)**
    - **brandingHeaderMidLTR.gif (652*1 pixel image)**
    - **brandingHeaderMidRTR.gif (652*1 pixel image)**
    - **ciscoLogo12pxMargin.gif (44*44 pixel image)**
    - **branding_logo.png (44*25 pixel image)**

### User Interface Branding Options

The following images display the customization options for the Cisco Unified CM Administration user interface:

*Figure 10: Branding Options for Unified CM Administration Login Screen*
The following table describes the callout options.

**Table 50: User Interface Branding Options: Login Screen**

<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
<th>Branding edits</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Company Logo</td>
<td>To add your logo to Cisco Unified Communications Manager, save your company logo as a 44x44 pixel image with the following filename: <code>ciscoLogol2pxMargin.gif</code> (44x44 pixels)</td>
</tr>
<tr>
<td></td>
<td></td>
<td><strong>Note</strong> If you also want to append your logo to the header and login screen of the Self-Care Portal, you should also save your logo as the 44x25 pixel <code>branding_logo.png</code> file.</td>
</tr>
<tr>
<td>2</td>
<td>Unified CM Administration header font color</td>
<td><code>heading.heading.color</code></td>
</tr>
<tr>
<td>Item</td>
<td>Description</td>
<td>Branding edits</td>
</tr>
<tr>
<td>------</td>
<td>------------------------------</td>
<td>-------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| 3    | Header Background            | You can use a single image or a combination of six images to create a grading effect.  
|      |                              | **Single Image option** — Save your header background as a single image:     |
|      |                              | • brandingHeader.gif (2048*1 pixel)                                          |
|      |                              | **Graded background option:** — Save your header background as six images for a graded effect: |
|      |                              | • brandingHeaderBegLTR.gif (652*1 pixel)                                    |
|      |                              | • brandingHeaderBegRTR.gif (652*1 pixel)                                    |
|      |                              | • brandingHeaderEndLTR.gif (652*1 pixel)                                    |
|      |                              | • brandingHeaderEndRTR.gif (652*1 pixel)                                    |
|      |                              | • brandingHeaderMidLTR.gif (652*1 pixel)                                    |
|      |                              | • brandingHeaderMidRTR.gif (652*1 pixel)                                    |
| 4    | Navigation text             | header.navigation.color                                                       |
| 5    | Go button                   | header.go.font.color                                                          |
|      |                              | header.go.background.color                                                    |
|      |                              | header.go.border.color                                                        |
| 6    | Username text               | splash.username.color                                                         |
| 7    | Password text               | splash.password.color                                                         |
| 8    | Login button                | splash.login.text.color                                                       |
|      |                              | splash.login.back.ground.color                                                |
| 9    | Reset button                | splash.reset.text.color                                                       |
|      |                              | splash.reset.back.ground.color                                                |
| 10   | Bottom background color – right | splash.hex.code.3                                                              |
| 11   | Bottom background color – left | splash.hex.code.2                                                             |
| 12   | Banner                      | splash.hex.code.1                                                             |

**Table 51: User Interface Branding Options: Logged In Screen**

<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
<th>Branding edits</th>
</tr>
</thead>
<tbody>
<tr>
<td>13</td>
<td>User text (for example, 'admin')</td>
<td>header.admin.color</td>
</tr>
<tr>
<td>14</td>
<td>Search, About and Login text</td>
<td>header.hover.link.color</td>
</tr>
</tbody>
</table>
**Branding Properties Editing Example**

Branding properties can be edited by adding the hex code in the properties file (BrandingProperties.properties). The properties file uses HTML-based hex code. For example, if you want to change the color of the Navigation text item (callout item #4) to red, add the following code to your properties file:

`header.navigation.color="#FF0000"`

In this code, `header.navigation.color` is the branding property that you want to edit, and "#FF0000" is the new setting (red).

---

<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
<th>Branding edits</th>
</tr>
</thead>
<tbody>
<tr>
<td>15</td>
<td>Unified CM Administration text heading</td>
<td>splash.header.color</td>
</tr>
<tr>
<td>16</td>
<td>System Version, VMware Installation text</td>
<td>splash.reset.text.color splash.version.color</td>
</tr>
</tbody>
</table>
Client Matter Codes and Forced Authorization Codes Overview

With client matter codes (CMCs) and forced authorization codes (FACs), you can effectively manage call access and accounting. CMCs assist with call accounting and billing for clients, and FACs regulate the types of calls that certain users can place.

CMCs force the user to enter a code; this action specifies that the call relates to a specific client matter. You can assign client matter codes to customers, students, or other populations for call accounting and billing purposes. FACs force the user to enter a valid authorization code that is assigned at a certain access level before the call is completed.

Client Matter Codes and Forced Authorization Codes Prerequisites

- Cisco Unified IP Phones that are running SCCP and SIP support CMC and FAC.
- The CMC and FAC tones play only on Cisco Unified IP Phones that are running SCCP or SIP; TAPI/JTAPI ports; and MGCP FXS ports.

Client Matter Codes and Forced Authorization Codes Configuration Task Flow

You can implement CMCs and FACs separately or together. For example, you may authorize users to place certain classes of calls, such as long distance calls, and also assign the class of calls to a specific client. CMC
and FAC tones sound the same to the user; if you configure both codes, the feature prompts the user to enter the FAC after the first tone and enter the CMC after the second tone.

Before you begin

- Review Client Matter Codes and Forced Authorization Codes Prerequisites, on page 593

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>After you finalize the list of CMCs that you plan to use, add those codes to the database and enable the CMC feature in route patterns.</td>
</tr>
<tr>
<td>To Configure Client Matter Codes, on page 594, complete the following subtasks:</td>
<td></td>
</tr>
<tr>
<td>• Add Client Matter Codes, on page 594</td>
<td></td>
</tr>
<tr>
<td>• Enable Client Matter Codes, on page 595</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong></td>
<td>After you finalize the list of FACs and authorization levels that you plan to use, add those codes to the database and enable the FAC feature in route patterns.</td>
</tr>
<tr>
<td>To Configure Forced Authorization Codes, on page 595, complete the following subtasks:</td>
<td></td>
</tr>
<tr>
<td>• Add Forced Authorization Codes, on page 596</td>
<td></td>
</tr>
<tr>
<td>• Enable Forced Authorization Codes, on page 596</td>
<td></td>
</tr>
</tbody>
</table>

Configure Client Matter Codes

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Determine unique client matter codes that you want to use and add them to your system. Because the number of CMCs directly affects the time that is required for your system to start up, limit the number of CMCs to a maximum of 60,000. If you configure more CMCs than the maximum number, expect significant delays.</td>
</tr>
<tr>
<td>Add Client Matter Codes, on page 594</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong></td>
<td>Enable client matter codes through a route pattern.</td>
</tr>
<tr>
<td>Enable Client Matter Codes, on page 595</td>
<td></td>
</tr>
</tbody>
</table>

Add Client Matter Codes

Determine unique client matter codes that you want to use and add them to your system. Because the number of CMCs directly affects the time that is required for your system to start up, limit the number of CMCs to a maximum of 60,000. If you configure more CMCs than the maximum number, expect significant delays.
Enable Client Matter Codes

Enable client matter codes through a route pattern.

Before you begin
Add Client Matter Codes, on page 594

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Route Pattern.
Step 2 Perform one of the following tasks:
   • To update an existing route pattern, enter search criteria, click Find, and choose a route pattern from the resulting list.
   • To create a new route pattern, click Add New.
Step 3 In the Route Pattern Configuration window, check the Require Client Matter Code check box.
Step 4 Click Save.

Configure Forced Authorization Codes

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add Forced Authorization Codes, on page 596 Determine unique forced authorization codes that you want to use and add them to your system.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Enable Forced Authorization Codes, on page 596 Enable forced authorization codes through a route pattern.</td>
</tr>
</tbody>
</table>
Add Forced Authorization Codes

Use this procedure to determine unique forced authorization codes that you want to use and add them to your system. To successfully route a call, the user authorization level must be equal to or greater than the authorization level that is specified for the route pattern for the call.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Forced Authorization Codes.</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>In the <strong>Authorization Code Name</strong> field, enter a unique name that is no more than 50 characters.</td>
</tr>
<tr>
<td></td>
<td>This name ties the authorization code to a specific user or group of users.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the <strong>Authorization Code</strong> field, enter a unique authorization code that is no more than 16 digits. Users enter this code when they place a call through an FAC-enabled route pattern.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Authorization Level</strong> field, enter a three-digit authorization level in the range of 0 to 255.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save.</strong></td>
</tr>
</tbody>
</table>

Enable Forced Authorization Codes

Use this procedure to enable forced authorization codes through a route pattern.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Call Routing &gt; Route/Hunt &gt; Route Pattern.</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Perform one of the following tasks:</td>
</tr>
<tr>
<td></td>
<td>• Click <strong>Find</strong>, and then choose a route pattern from the resulting list to update an existing route pattern.</td>
</tr>
<tr>
<td></td>
<td>• Click <strong>Add New</strong> to create a new route pattern.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the <strong>Route Pattern Configuration</strong> window, check the <strong>Require Forced Authorization Code</strong> check box.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Authorization Level</strong> field, enter the authorization level value between 0 and 255. The FAC level for the user must be greater than or equal to the configured level for the call to route successfully.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save.</strong></td>
</tr>
</tbody>
</table>

Client Matter Codes and Forced Authorization Codes Interactions and Restrictions

- Client Matter Codes and Forced Authorization Codes Interactions, on page 597
- Client Matter Codes and Forced Authorization Codes Restrictions, on page 598
# Client Matter Codes and Forced Authorization Codes Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>CDR Analysis and Reporting (CAR)</td>
<td>CDR Analysis and Reporting (CAR) allows you to run reports that provide call details for client matter codes (CMCs), forced authorization codes (FACs), and authorization levels.</td>
</tr>
</tbody>
</table>
| CTI, JTAPI, and TAPI applications | In most cases, your system can alert a CTI, JTAPI, or TAPI application that the user must enter a code during a call. When a user places a call, creates an ad hoc conference, or performs a consult transfer through a CMC- or FAC-enabled route pattern, the user must enter a code after receiving the tone.  
When a user redirects or blind transfers a call through a CMC- or FAC-enabled route pattern, the user receives no tone, so the application must send the codes to Cisco Unified Communications Manager. If your system receives the appropriate codes, the call connects to the intended party. If your system does not receive the appropriate codes, Cisco Unified Communications Manager sends an error to the application that indicates which code is missing. |
| Cisco Web Dialer           | Web Dialer supports CMCs and FACs in the following ways:                                                                                                                                                      |
|                           | • A user can enter the destination number in the dial text box of the WD HTML page or SOAP request, and then manually enter the CMC or FAC on the phone.                                                        |
|                           | • A user can enter the destination number followed by the FAC or CMC in the dial text box of the WD HTML page or SOAP request.                                                                                 |
|                           | For example, if the destination number is 5555, the FAC is 111, and the CMC is 222, a user can make a call by dialing 5555111# (FAC), 5555222# (CMC), or 5555111222# (CMC and FAC).                                           |
|                           | **Note**                                                                                                                                                                                                    |
|                           | • WebDialer does not handle any validation for the destination number. The phone handles the required validation.                                                                                          |
|                           | • If a user does not provide a code or provides the wrong code, the call will fail.                                                                                                                         |
|                           | • If a user makes a call from the WebApp with a DN that contains special characters, the call goes successfully after stripping the special characters. The same rules do not work in SOAP UI.               |
| Speed Dial and Abbreviated Speed Dial | You can use speed dial to reach destinations that require a FAC, CMC, dialing pauses, or additional digits (such as a user extension, a meeting access code, or a voicemail password). When the user presses the configured speed dial, the phone establishes the call to the destination number and sends the specified FAC, CMC, and additional digits with dialing pauses inserted. |
# Client Matter Codes and Forced Authorization Codes Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Analog gateways</td>
<td>H.323 analog gateways do not support CMCs or FACs because these gateways cannot play tones.</td>
</tr>
<tr>
<td>Call forwarding</td>
<td>Calls that are forwarded to a CMC- or FAC-enabled route pattern fail because no user is present to enter the code. When a user presses the CFwdALL softkey and enters a number that has CMC or FAC enabled on the route pattern, call forwarding fails. To minimize call-processing interruptions, test the number before you configure call forwarding. To do this, dial the intended forwarding number; if you are prompted for a code, do not configure call forwarding for that number. Advise users of this practice to reduce the number of complaints that result from forwarded calls that do not reach the intended destination.</td>
</tr>
<tr>
<td>Cisco Unified Mobility</td>
<td>Calls that originate from a SIP trunk, H.323 gateway, or MGCP gateway fail if they encounter a route pattern that requires CMCs or FACs and the caller is not configured with Cisco Unified Mobility.</td>
</tr>
<tr>
<td>Dial via Office callback number</td>
<td>The CMC and FAC feature on Cisco Mobility does not support an alternative number as its dial via office (DVO) callback number. The DVO callback number must be the number that is registered on the Mobility Identity window.</td>
</tr>
<tr>
<td>Failover calls</td>
<td>CMCs and FACs do not work with failover calls.</td>
</tr>
<tr>
<td>Hearing-impaired users</td>
<td>After dialing the phone number, hearing-impaired users should wait one or two seconds before entering the authorization or client matter code.</td>
</tr>
<tr>
<td>Localization</td>
<td>Cisco does not localize CMCs or FACs. The CMC and FAC features use the same default tone for any locale that is supported with Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Note</td>
<td>For Cisco Mobility, CMCs and FACs are localized.</td>
</tr>
<tr>
<td>Overlap sending</td>
<td>The CMC and FAC features do not support overlap sending because Cisco Unified Communications Manager cannot determine when to prompt the user for the code. If you check the Require Forced Authorization Code or the Require Client Matter Code check box in the Route Pattern Configuration window, the Allow Overlap Sending check box is automatically unchecked and vice-versa.</td>
</tr>
<tr>
<td>Speed-dial buttons</td>
<td>You cannot configure CMCs or FACs for speed-dial buttons. You must enter the code when the system prompts you to do so.</td>
</tr>
</tbody>
</table>
Custom Phone Rings and Backgrounds

- Custom Phone Rings Overview, on page 599
- Custom Phone Rings Prerequisites, on page 599
- Custom Phone Rings Configuration Task Flow, on page 600
- Custom Backgrounds, on page 602
- Custom Backgrounds Configuration Task Flow, on page 603

Custom Phone Rings Overview

Custom Phone Rings allows you to create customized phone rings and upload the customized files to the Cisco Unified Communications Manager TFTP server where they can be accessed by Cisco Unified IP Phones.

Cisco Unified IP Phones ship with default ring types that are implemented in hardware: Chirp1 and Chirp2. In addition, Cisco Unified Communications Manager provides the capability of uploading the following files to phones:

**PCM Files**

Cisco Unified Communications Manager provides a default set of phone ring sounds that are implemented in software as pulse code modulation (PCM) audio files. Each PCM file specifies a single ring type.

**Ringlist.xml File**

The Ringlist.xml file describes the list of ring options that are available for phones.

You can upload customized PCM audio files, such as custom ring tones and callback tones, as well as the modified Ringlist.xml file to the TFTP directory in Cisco Unified Communications Manager.

Custom Phone Rings Prerequisites

The following prerequisites apply to Custom Phone Rings:

- In order to upload your custom phone rings, the Cisco TFTP service must be running.
- Any PCM files that you want to upload must meet a set of file requirements in order to be compatible with Cisco Unified IP Phones. For details, review the topic PCM File Format Requirements, on page 601.
- The Ringlist.xml file must meet a set of formatting guidelines. For details, review the topic Ringlist.xml File Format Requirements, on page 601.
Custom Phone Rings Configuration Task Flow

Before you begin
- Review Custom Phone Rings Prerequisites, on page 599

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Prepare Custom Phone Rings for Upload, on page 600</td>
<td>Create your customized PCM and Ringlist.xml files.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Upload Custom Phone Rings to TFTP Server, on page 600</td>
<td>Upload customized files to the Cisco Unified Communications Manager TFTP server.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Restart TFTP Service, on page 601</td>
<td>After the upload completes, restart the Cisco TFTP service.</td>
</tr>
</tbody>
</table>

Prepare Custom Phone Rings for Upload

Procedure

**Step 1** Use the `file get tftp <tftp path>` CLI command to download the existing Ringlist.xml file, in addition to any PCM files that you want to modify.

**Step 2** Create a PCM file for each ring type that you want to upload. For guidelines on PCM file compatibility with Cisco Unified Communications Manager, see PCM File Format Requirements, on page 601.

**Step 3** Use an ASCII editor to update the Ringlist.xml file with your new phone rings. For details on Ringlist.xml file formatting requirements, see Ringlist.xml File Format Requirements, on page 601.

What to do next
Upload Custom Phone Rings to TFTP Server, on page 600

Upload Custom Phone Rings to TFTP Server

**Before you begin**
Prepare Custom Phone Rings for Upload, on page 600

**Procedure**

**Step 1** In Cisco Unified OS Administration, choose Software Upgrades > TFTP > File Management.
**Step 2** Click **Upload File**.

**Step 3** Click **Browse** and select the Ringlist.xml file, as well as any PCM files that you want to upload.

**Step 4** Click **Upload File**.

---

**What to do next**

Restart TFTP Service, on page 601

---

**Restart TFTP Service**

**Before you begin**

Upload Custom Phone Rings to TFTP Server, on page 600

**Procedure**

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

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

**Step 3** Click the radio button that corresponds to the **Cisco TFTP** service.

**Step 4** Click **Restart**.

---

**PCM File Format Requirements**

PCM files for phone rings must meet a set of requirements for proper playback on Cisco Unified IP Phones. When creating or modifying your PCM files, you can use any standard audio editing packages that support the following file format requirements:

- Raw PCM
- 8000 samples per second
- 8 bits per sample
- mu-law compression
- Maximum ring size: 16080 samples
- Number of samples in the ring must be evenly divisible by 240
- Ring starts and ends at the zero crossing

---

**Ringlist.xml File Format Requirements**

The Ringlist.xml file defines an XML object that contains a list of phone ring types. Each ring type contains a pointer to the PCM file that is used for that ring type and the text that will display on the Ring Type menu on a Cisco Unified IP Phone for that ring.

The CiscoIPPhoneRinglist XML object uses the following simple tag set to describe the information:

```xml
<CiscoIPPhoneRinglist>
  <Ring>
    <DisplayName/>
  </Ring>
</CiscoIPPhoneRinglist>
```
The following characteristics apply to the definition names:

- **DisplayName** defines the name of the custom ring for the associated PCM file that will display on the Ring Type menu of the Cisco Unified IP Phone.
- **FileName** specifies the name of the PCM file for the custom ring to associate with **DisplayName**.

**Tip**
The **DisplayName** and **FileName** fields must not exceed 25 characters.

The following example shows a Ringlist.xml file that defines two phone ring types:

```xml
<CiscoIPPhoneRinglist>
  <Ring>
    <DisplayName>Analog Synth 1</DisplayName>
    <FileName>Analog1.raw</FileName>
  </Ring>
  <Ring>
    <DisplayName>Analog Synth 2</DisplayName>
    <FileName>Analog2.raw</FileName>
  </Ring>
</CiscoIPPhoneRinglist>
```

**Tip**
You must include the required **DisplayName** and **FileName** for each phone ring type. The Ringlist.xml file can include up to 50 ring types.

---

## Custom Backgrounds

You can also use the TFTP server to upload new custom background images to the phones in your network. Phone users can select the images that you upload as their phone backgrounds. You can configure your system so that phone users can select from an assortment of images or you can assign a specific background image for all phones.

If you want your phone users to be able to customize their phone backgrounds, then you must prepare and upload the following files to the TFTP server whenever you upload new images:

- Full-size background image—Refer to your phone documentation for image specifications, including the image size (in pixels) and color-type, for your phone model.
- A thumbnail image—This is only required if you want your phone users to be able to choose their own background image. Refer to your phone documentation for the thumbnail image specifications
- An edited **List.xml** file—This file contains a listing of the background images from which phone users can select. You must add your new images to this file.

If you want to assign a specific image for all phones then you need to upload the main background image only. In addition, you also must update the Common Phone Profile to direct the phones to use the image that you assign.
## Custom Backgrounds Configuration Task Flow

Complete these tasks to configure and upload customized background images for the phones in your deployment. You can configure the system so that phone users can select from an assortment of images, or you can assign a specific background image that displays on all phones.

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Create Phone Background Images, on page 604</td>
<td>Create your full-size background image and corresponding thumbnail image (if required). Refer to your phone documentation for image specifications, including file type, image size (in pixels) and color-type. <strong>Note</strong> The thumbnail is not required if you are assigning a specific background image.</td>
</tr>
<tr>
<td>2</td>
<td>Edit the List.xml file, on page 604</td>
<td>Update the List.xml file from the appropriate TFTP directory with your new images. This is required so that phone users see the new images in their list of phone background options. <strong>Note</strong> This procedure is required only if you are giving your users the option to choose their own background. If you are assigning a specific background image then there is no need to edit this file.</td>
</tr>
<tr>
<td>3</td>
<td>Upload Backgrounds to TFTP Server, on page 605</td>
<td>Upload your files to the TFTP server.</td>
</tr>
<tr>
<td>4</td>
<td>Restart the TFTP Server, on page 605</td>
<td>Restart the Cisco TFTP service in order to push the images to your phones.</td>
</tr>
<tr>
<td>5</td>
<td>Assign Phone Background for Phone Users, on page 606</td>
<td>Optional. By default, Cisco Unified Communications Manager gives phone users the option to select their own phone background image. However, you can use the Common Phone Profile to assign a specific background image for all phones that use this Common Phone Profile.</td>
</tr>
</tbody>
</table>
Create Phone Background Images

Refer to your phone documentation for background image specifications and thumbnail image specifications. This includes the image sizes (in pixels), file type, and the appropriate destination TFTP directory for that phone model (the TFTP directory is based on the image specifications).

- If you want phone users to have the option to use, or not use, the image that you upload, you must prepare both a full-size image and a thumbnail image according to the specifications for that particular phone model.
- If you want to assign the image to specific phones, you need the full-size image only.

What to do next

If you want phone users to be able to choose their own background image, Edit the List.xml file, on page 604.
If you want to assign a specific background image, you don't need to update the List.xml file. Proceed to Upload Backgrounds to TFTP Server, on page 605

Edit the List.xml file

If you want phone users to be able to choose their background images, use this procedure to add any new background images that you upload to the existing List.xml file. Each TFTP image directory contains a List.xml file that gets used by the phones that use that TFTP directory. This file points to the specific background and thumbnail image for each background option and can include up to 50 background images. The images are listed using the order in which they appear on the phone. For each image, the file contains an <ImageItem> element that includes these two attributes:

- Image: Uniform resource identifier (URI) that specifies where the phone obtains the thumbnail image that will appear on the Background Images menu of a phone.
- URL: URI that specifies where the phone obtains the full size image.

Example:
The following example (for a Cisco Unified IP Phone 7971G-GE and 7970G) shows a List.xml file that defines two images. The required Image and URL attributes must be included for each image. The TFTP URI that displays in the example is the only supported method for linking to full size and thumbnail images as HTTP URL support is not provided.

```
<CiscoIPPhoneImageList>
  <ImageItem Image="TFTP:Desktops/320x212x12/TN-Fountain.png" URL="TFTP:Desktops/320x212x12/Fountain.png"/>
  <ImageItem Image="TFTP:Desktops/320x212x12/TN-FullMoon.png" URL="TFTP:Desktops/320x212x12/FullMoon.png"/>
</CiscoIPPhoneImageList>
```

Procedure

**Step 1** Log in to the Command Line Interface

**Step 2** Run the `file get tftp <filename>` CLI command where `<filename>` represents the file and filepath of the List.xml file for the appropriate TFTP directory.
Note Make sure that you download the List.xml file from the appropriate TFTP directory as each image directory has its own file. Refer to your phone documentation for information on the appropriate TFTP directory for that phone model as the directory is based on the image specifications.

Step 3 Edit the xml file with a new <ImageItem> element for each new background option that you want to add.

What to do next
Upload Backgrounds to TFTP Server, on page 605

Upload Backgrounds to TFTP Server

Use this procedure to upload new phone background files to the TFTP server.

- If you want your phone users to be able to choose their own background image, then you must upload your full-size background image, a thumbnail image, and the updated List.xml file.
- If you are assigning a specific background image, you need to upload the full-size background image only.

Procedure

Step 1 From Cisco Unified OS Administration, choose Software Upgrades > TFTP File Management
Step 2 Click Upload File and do the following:
  a) Click Choose File and select the background file that you want to upload.
  b) In the Directory field, enter the appropriate TFTP directory for that phone model. The TFTP directory corresponds to the image size and color type. Refer to your phone documentation for images specification.
  c) Click Upload File
  d) Repeat these steps to upload both the thumbnail image and list.xml files as well. These files should be loaded to the same TFTP directory as the main background image.

Step 3 Click Close.

What to do next
Restart the TFTP Server, on page 605

Restart the TFTP Server

After you have uploaded your custom files to the TFTP directory, restart the Cisco TFTP server to push the files to the phones.

Procedure

Step 1 Log in to Cisco Unified Serviceability and choose Tools > Control Center - Feature Services.
Step 2  From the Server drop-down list box, choose the server on which the Cisco TFTP service is running.
Step 3  Click the radio button that corresponds to the Cisco TFTP service.
Step 4  Click Restart.

What to do next
Assign Phone Background for Phone Users, on page 606

Assign Phone Background for Phone Users

By default, Cisco Unified Communications Manager allows phone users to customize their own phone background image. However, you can use the Common Phone Profile setting to assign a specific background image for all phones that use this Common Phone Profile.

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Device Settings > Common Phone Profile.
Step 2  Do one of the following:
   • Click Find and select the Common Phone Profile that your phones use.
   • Click Add New to create a new Common Phone Profile.
Step 3  If you want users to be able to choose their background image, make sure that the Enable End User Access to Phone Background Image Setting check box is checked (this is the default setting).
Step 4  If you want to assign a specific background image for phones that use this profile:
   • Uncheck the Enable End User Access to Phone Background Image Setting check box.
   • In the Background Image text box, enter the filename of the image file that you want to assign. Also, check the Override Enterprise Settings check box that corresponds to this text box.
Step 5  Complete any remaining fields in the Common Phone Profile window. For help with the fields and their settings, refer to the online help.
Step 6  Click Save.
If you have assigned a specific background image, all phones that use this Common Phone Profile will use the image that you specify.

What to do next
If you have created a new Common Phone Profile, reconfigure your phones so that they use this profile. For details on how to configure phones in Cisco Unified Communications Manager, see the “Configure Endpoint Devices” section of the System Configuration Guide for Cisco Unified Communications Manager.

Tip
If you have a large number of phones to assign, use the Bulk Administration Tool to assign a Common Phone Profile to a large number of phones in a single operation. For details, see the Bulk Administration Guide for Cisco Unified Communications Manager.
After you complete your configuration, reset your phones.
Assign Phone Background for Phone Users
Music On Hold Overview

Use the integrated Music On Hold (MOH) feature to place on-net and off-net users on hold with music from a streaming source. This source makes music available to any on-net or off-net device that you place on hold. On-net devices include station devices and applications that an interactive voice response (IVR) or call distributor places on hold, consult hold, or park hold. Off-net users include those users who are connected through Media Gateway Control Protocol (MGCP) or Skinny Call Control Protocol (SCCP) gateways, Cisco IOS H.323 gateways, and Cisco IOS Media Gateway Control Protocol gateways. The system also makes the Music On Hold feature available for Cisco IP POTS phones that connect to the Cisco IP network through Foreign Exchange Station (FXS) ports on Cisco IOS H.323 or MGCP and for Cisco MGCP or SCCP gateways.

Start Cisco Unified Communications Manager to create a media resource manager. Music On Hold server registers to the media resource manager with its music on hold resources. Music On Hold server is a software application that provides music on hold audio sources and connects a music on hold audio source to multiple streams.

When an end device or feature places a call on hold, Cisco Unified Communications Manager connects the held device to a music resource. When the held device is retrieved, it disconnects from the music on hold resource and resumes normal activity.

Caller-Specific Music On Hold

For SIP calls that a phone receives over the SIP trunk, Cisco Unified Communications Manager can use a different MOH audio source.

An external application, such as the Cisco Unified Customer Voice Portal (CVP) contact center solution, determines the most appropriate MOH audio source based on the caller ID, dialed number, or IVR interaction when a call is received from the public switched telephone network (PSTN).
Increased Capacity of IP Voice Media Streaming Application and Expanded MOH Audio Source

Cisco IP Voice Media Streaming application is installed automatically when you install Cisco Unified Communications Manager. Activate this application to enable the Music On Hold (MOH) feature.

With this release, the capacity of Cisco Unified Communications Manager to support unique and concurrent MOH audio sources, while the Music On Hold service is running on the MOH server, is increased from 51 to 501. The MOH audio sources are numbered from 1 to 501 with the fixed MOH audio source remaining at the number 51.

The fixed MOH device cannot use an audio source that connects through a USB MOH device, because Cisco Unified Communications Manager does not support USB when running on VMware. Use of the fixed MOH USB device is not supported on VMware. However, provision the external sound device for use with deployments that utilize Cisco Unified Survivable Remote Site Telephony (SRST) multicast MOH.

You can configure each MOH audio source to use a custom announcement as an initial greeting and/or an announcement that is played periodically to callers who are hearing the music. Cisco Unified Communications Manager provides 500 custom announcementsthat you can use on one or multiple MOH audio sources. These announcements are not distributed between the Cisco Unified Communications Manager servers within a cluster. You have to upload these custom announcement files to each server that provides the MOH and announcement services. You must also upload each custom music file for MOH audio sources to each server.

Performance Impact of Media Devices with Services

The Cisco IP Voice Media Streaming application runs as a service for four media devices—annunciator (ANN), software conference bridge, Music On Hold (MOH), and software media termination point. Activate this service on a Cisco Unified Communications Manager server as coresident with call processing. When you activate this service, ensure that you configure these media devices for limited capacity to avoid any impact on the call processing. The default settings for the media devices are defined based on this coresident operation. You can adjust these settings by reducing the use of one or more media devices to increase other settings.

For example, if you are not using software media termination point devices, you can choose the Run Flag setting for the SW MTP to False, select System > Service Parameters > Cisco IP Voice Media Streaming App service > MTP Parameters, and add the MTP Call Count setting to Media Resource > MOH Server > Maximum Half Duplex Streams configuration. Depending on the call traffic, you can modify the default settings. However, monitor the server performance activity for CPU, memory, and IO wait. For higher capacity clusters, such as the ones using 7500 user OVA configuration, it is possible to increase the default media device settings for Call Count by 25%.

For installations where you expect high usage of the media devices, such as Music On Hold, or where high call volumes require higher number of media connections, activate the Cisco IP Voice Media Streaming application service on one or more of the Cisco Unified Communications Manager servers which do not have call processing activated. Activating this service limits the impact of media device usage to other services, such as call processing. Then, you can increase the configuration settings for maximum number of calls for the media devices.

When you activate Cisco IP Voice Media Streaming application as co-resident with Cisco Unified Communications Manager service, it can impact call processing performance. To increase the capacity settings
for Music On Hold or announciator from the default settings, it is suggested to activate Cisco IP Voice Media Streaming application on a server without activating Cisco Unified Communications Manager. The CPU performance is impacted by MOH when active callers are on hold or when multicast MOH audio streams are configured.

**Table 54: General Performance Results**

<table>
<thead>
<tr>
<th>Configuration Notes</th>
<th>CPU Performance</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dedicated MOH server, 1000 held calls, 500 MOH sources with greeting and periodic announcements.</td>
<td>25–45% (7500 user OVA configuration)</td>
</tr>
<tr>
<td>Native call queuing with dedicated MOH server and announciator server, 1000 queued calls, 500 MOH sources with greeting and periodic announcements. An announciator can play up to 300 simultaneous greeting announcements.</td>
<td>25–45% (7500 user OVA configuration)</td>
</tr>
<tr>
<td>Dedicated MOH server, 500 held calls, 500 MOH sources with greeting and periodic announcements.</td>
<td>15–35% (7500 user OVA configuration)</td>
</tr>
</tbody>
</table>

**Table 55: Extrapolated Recommendations**

<table>
<thead>
<tr>
<th>Configuration</th>
<th>Recommendation Limit</th>
</tr>
</thead>
<tbody>
<tr>
<td>When Cisco IP Voice Media Streaming application is co-resident with Cisco Unified Communications Manager on 2500 OVA (moderate call processing).</td>
<td>MOH: 500 held callers, 100 MOH sources, and 48 to 64 announciator callers.</td>
</tr>
<tr>
<td>When Cisco IP Voice Media Streaming application is a dedicated server on 2500 OVA.</td>
<td>MOH: 750 held callers, 250 MOH sources, and 250 announciator callers.</td>
</tr>
<tr>
<td>When Cisco IP Voice Media Streaming application is co-resident with Cisco Unified Communications Manager on 7500/10K OVA (moderate call processing).</td>
<td>MOH: 500 held callers, 250 MOH sources, and 128 announciator callers.</td>
</tr>
<tr>
<td>When Cisco IP Voice Media Streaming application is a dedicated server on 7500/10K OVA.</td>
<td>MOH: 1000 held callers, 500 MOH sources, and 300-700 announciator callers (with 1 MOH codec). Note Reduce announciator to 300 for two MOH codecs.</td>
</tr>
</tbody>
</table>

These recommendations are specific to MOH/ANN devices. If you combine these devices with the software media termination point (MTP) and call forward busy (CFB) devices, reduce the limits to provide streams.
Configuration Limitations for Capacity Planning

The Cisco IP Voice Media Streaming application and Self Provisioning IVR services use a media kernel driver to create and control Real-time Transfer Protocol (RTP) streams. This media kernel driver has a capacity of 6000 streams. These streams allow the media devices and IVR to make resource reservations.

These reservations are based on the following capacity calculations:

<table>
<thead>
<tr>
<th>Media Device</th>
<th>Capacity</th>
</tr>
</thead>
</table>
| Annunciator                        | (Call Count service parameter) * 3  
(Where 3 indicates total of receiving (RX) and transmitting (TX) calls for endpoint and 1 for .wav file.) |
| Software Conference Bridge         | (Call Count service parameter) * 2  
(Where 2 indicates total streams of RX and TX endpoints.) |
| Software Media Termination Point   | (Call Count service parameter) * 2  
(Where 2 indicates total streams of RX and TX endpoints.) |
| Music On Hold                      | (Maximum Half Duplex Streams) * 3) + (501 * 2 * [number of enabled MOH codecs])  
(Where:  
• (Maximum Half Duplex Streams) is a configuration setting on the MOH device configuration administration web page.  
• 3 indicates total streams of RX, TX, and greeting announcement .wav file.  
• 501 indicates the maximum number of Music On Hold (MOH) sources.  
• 2 indicates music .wav stream and possible multicast TX stream.  
• [number of enabled MOH codecs] is based on how many MOH codecs are enabled in the Cisco IP Voice Media Streaming application service parameters.) |
| Self Provisioning IVR Service      | (500 * 2)  
(Where 500 indicates callers, and 2 indicates total streams from RX and TX streams.) |

Hence, to enable MOH to support a maximum of 1000 callers, use the following equation: 1000 * 3 + 501 * 2 * 1 = 4002 driver streams with one enabled codec and 1000 * 3 + 501 * 2 * 2 = 5004 with two enabled codecs. Reduce the remaining devices and deactivate the Self Provisioning IVR service to limit total reservations to 6000, which allows the MOH device to make these reservations. It may also require that you do not activate the Self Provisioning IVR service on the same server with Cisco IP Voice Media Streaming application.

If configuration settings of the media devices exceed the capacity of the media device driver, the media devices that register with the device driver first will be able to reserve their required stream resources. The media devices that register later are restricted to fewer than requested stream resources. The later registered media
devices result in logging some alarm messages and automatically reducing the call count for the restricted media device.

---

Note

A media kernel driver with a capacity of 6000 streams might not support that many simultaneous media device connections.

---

**Interwork External Multicast MOH to Unicast MOH**

With this release, you can configure a Cisco Unified Survivable Remote Site Telephony (SRST) router as an audio source. This router provides multicast MOH audio for devices that are capable of multicast reception. In this approach, devices act as if Cisco Unified Communications Manager is sending the multicast MOH audio. However, devices that are capable of only the unicast reception cannot hear the MOH audio that an external MOH source (for example, Cisco Unified SRST router) sends. Examples of devices that are capable of unicast reception only can be public switched telephone network (PSTN) phones, destination to session border controllers (SBC), and Session Initiation Protocol (SIP) trunks.

In this release of Cisco Unified Communications Manager, this feature is enhanced to receive multicast MOH audio from an external audio source and send it as unicast MOH audio. Cisco Unified Communications Manager uses this feature to play multicast MOH audio as unicast MOH for the devices that are capable of unicast MOH reception only. Examples of an external MOH audio source can be a Cisco Unified SRST router or software that can send multicast MOH audio.

An administrator configures the fields for this feature from Cisco Unified CM Administration *Music On Hold Audio Source Configuration* window.

---

Note

- This feature has no impact on existing functionality of playing multicast MOH audio using an external audio source for the devices that are capable of multicast reception.

- For the unicast media connection, Cisco Unified Communications Manager MOH Server plays the initial announcement and periodic announcement even if you configure the MOH audio source with external multicast source.

---

**Configuration Tips for the Codec-Specific Inbound Audio Stream**

Configure an external multicast audio source, such as Cisco Unified SRST router, to MOH server for streaming the required audio feed.

To configure an external multicast audio source, such as a Cisco Unified SRST router, configure the *Source IPv4 Multicast Address* and *Source Port Number* fields in the *MOH Audio Source Configuration* window.

- Cisco Unified Communications Manager listens to multicast G.711 mu-law stream on external multicast IP address and port that you configured on the *MOH Audio Source configuration* window. An MOH server can transcode between the G.711 mu-law or a-law or L16 256K wideband MOH codecs. The external multicast RTP stream uses G.711 mu-law codec for MOH as a source for G.711 mu-law or a-law or L16 256K wideband MOH codecs. For G.711 a-law and wideband calls, Cisco Unified Communications Manager MOH server transcodes the inbound G.711 mu-law stream to outbound G.711 a-law or wideband stream before sending it to the device.
• Cisco Unified Communications Manager listens to multicast G.729 stream on external multicast IP and port value added with four that is configured on the MOH audio source configuration window. For example, if you configure an MOH audio Source with 239.1.1.1:16384, Cisco Unified Communications Manager listens to G.711 mu-law stream on 239.1.1.1:16384 and G.729 stream on 239.1.1.1:16388 (port value added with four). An MOH server cannot transcode for G.729 codecs. Callers who are using MOH G.729 codec require an external multicast RTP stream using G.729 or G.729a codec.

Music On Hold Prerequisites

• Before you configure multicast, ensure that you configure MOH server and audio sources. If you want to use fixed audio source, configure it before you configure multicast.

• Make sure to decide whether you are going to do unicast or multicast Music On Hold

• It is crucial to plan the capacity of the deployed and configured hardware and ensure that it can support the anticipated call volume of the network. You need to know the hardware capacity for MOH resources and consider the implications of multicast and unicast MOH in relation to this capacity. Ensure that network call volumes do not exceed these limits. When MOH sessions reach these limits, an additional load can result in poor MOH quality, erratic MOH operation, or loss of MOH functionality.

• If you use multicast MOH and the devices that listen to multicast MOH streams are not in the same IP network, you must enable multicast routing in the IP network. Take care when you enable the multicast routing to avoid the potential flooding of parts of the network with wrongly sent multicast packets (specially, across WAN links). Disable multicastrons on interfaces on which the multicast MOH packets are not required and use the Max Hops parameter.

• For detailed information on planning your Music On Hold deployment, including server capacities, refer to the Music On Hold capacities topics in the Cisco Collaboration System Solution Reference Network Design.

Music On Hold Configuration Task Flow

Complete these tasks to configure Music On Hold (MOH) for your system.

<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 Cisco IP Voice Media Streaming, on page 615</td>
<td>Activate the Cisco IP Voice Media Streaming Service Application service to enable Music On Hold.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Music On Hold Server, on page 616</td>
<td>Configure basic server settings for the MOH server.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Upload Audio File for Music On Hold, on page 616</td>
<td>Optional. Upload your own audio files to make them available as MOH audio streams.</td>
</tr>
</tbody>
</table>
Activate Cisco IP Voice Media Streaming

The Cisco IP Voice Media Streaming Application service must be Activated in order to use Music On Hold.

Note
During installation, Cisco Unified Communications Manager installs and configures a default Music On Hold audio source. Music On Hold functionality can proceed by using the default audio source.

Procedure

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

**Step 2** Choose a server from the Server drop-down list.

**Step 3** Under CM Services, make sure the Cisco IP Voice Media Streaming App service is Activated. If the service is deactivated, check the service and click Save.

What to do next
Configure Music On Hold Server, on page 616
Configure Music On Hold Server

Before you begin
Make sure one or multiple Music On Hold (MOH) servers are available.

Note
The Cisco Unified Communications Manager MOH server is automatically added when the Cisco IP Voice Media Streaming Application service is activated.

Procedure

Step 1
From Cisco Unified CM Administration, choose Media Resources > Music On Hold Server.

Step 2
Click Find and select the Music On Hold server that you want to update.

Step 3
Select the Host Server.

Step 4
Enter a descriptive Music On Hold Server Name along with a Description.

Step 5
Select the Device Pool you want to use for this server.

Step 6
Configure server capacity by configuring the following fields:

- **Maximum Half Duplex Stream**—Set this to the maximum number of devices that can be on unicast music on hold that is streamed from this music on hold server at any given time. You can use the following formula to calculate the maximum:

  \[ \text{Maximum Half Duplex Stream} = (\text{Server and deployment capacity}) - (\text{Number of multicast MOH sources} \times \text{Number of enabled MOH codecs}) \]

  Note

- **Maximum Multi-cast Connections**—Set this to a value that is greater than or equal to the number of devices that might be placed on multicast MOH at any given time.

Step 7
Optional. To enable multi-casting, check the Enable Multi-cast Audio Sources on this MOH Server check box, and configure the multicast IP address ranges.

Step 8
Configure the additional fields in the Music On Hold Server Configuration window. For help with the fields and their settings, see the online help.

Step 9
Click Save.

What to do next
Upload Audio File for Music On Hold, on page 616

Upload Audio File for Music On Hold

Use this procedure if you want to upload customized audio files that you can make available as Music On Hold audio streams.
Procedure

Step 1  From Cisco Unified CM Administration, choose Media Resources > MOH Audio File Management.
Step 2  Click Upload File.
Step 3  Click Choose File and browse to the file you want to upload. Once you've selected the file, click Open.
Step 4  Click Upload.

The Upload Result window shows the result of the upload. The uploading procedure uploads the file 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.

Step 5  Click Close to close the Upload Result window.
Step 6  Repeat this procedure if you want to upload additional audio files.

Note  When you import an audio source file, Cisco Unified Communications Manager processes the file and converts the file to the proper formats for use by the Music On Hold server. Following are examples of valid input audio source files:

- 16-bit PCM .wav file
- Stereo or mono
- Sample rates of 48 kHz, 44.1 kHz, 32 kHz, 16 kHz, or 8 kHz

Note  MOH audio source files do not automatically propagate to other MOH servers in a cluster. You must upload an audio source file to each MOH server or each server in a cluster separately.

What to do next
Configure Music On Hold Audio Source, on page 617

Configure Music On Hold Audio Source

Use this procedure to configure Music On Hold audio sources. You can configure audio streams and associate uploaded files to an audio stream. You can configure up to 500 audio streams.

Note  If a new version of an audio source file is available, perform the update procedure to use the new version.

Procedure

Step 1  From Cisco Unified CM Administration, choose Media Resources > Music On Hold Audio Source.
Step 2  Do either of the following:

- Click Find and select an existing audio stream.
- Click Add New to configure a new stream.
Configure Fixed Music On Hold Audio Source

For each cluster, you may define one fixed audio source (Source 51). You must set up the fixed audio source that is configured per cluster on each MOH server. The fixed audio source originates from a fixed device that uses the local computer audio driver.

**Procedure**

**Step 1** In the Cisco Unified Communications Manager Administration, choose Media Resources > Fixed MOH Audio Source.

**Step 2** Optional. Check the Allow Multi-casting check box if you want to allow this audio source to be multi-casted.

**Step 3** Check the Enable check box to enable the fixed audio source. When you check this check box, a Name is required.

**Step 4** In the Announcement Settings for Held and Hunt Pilot Calls area, configure announcements for this audio source.

**Step 5** Configure the fields in the Fixed MOH Audio Source Configuration window. For help with the fields and their settings, see the online help.

**Step 6** Click Save.

**What to do next**

Add MOH to Media Resource Group, on page 618

**Add MOH to Media Resource Group**

A Media Resource Group is a logical grouping of media resources. You may associate a media resource group with a geographical location or a site, as required. You can also form media resource groups to control server usage, or unicast or multicast service type.
Procedure

**Step 1**  In the Cisco Unified CM Administration, choose *Media Resources > Media Resource Group.*

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

**Step 3**  Enter a **Name** and **Description.**

**Step 4**  In the *Available Media Resources* list, select the Music On Hold resource and use the down arrow to add the resource to the *Selected Media Resources.* Repeat this step for the other media resources you want to assign to this group.

**Step 5**  Optional. Check the **Use Multi-cast for MOH Audio** check box if you want to allow Music On Hold multi-casting.

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

---

**What to do next**

Configure Media Resource Group List, on page 619

---

**Configure Media Resource Group List**

Media Resource Group List lists the prioritized media resource groups. An application can select required media resources from among ones that are available according to the priority order that is defined in a media resource group list.

**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 media resource group list.
- Click **Add New** to create a new media resource group list.

**Step 3**  Enter a **Name** for the list.

**Step 4**  From the *Available Media Resource Groups* list, select the groups you want to add to this list and use the down arrow to move them to *Selected Media Resource Groups.*

**Step 5**  In the *Selected Media Resource Groups* list use the up and down arrows to the right of the list to edit the prioritized order of groups.

**Step 6**  Click **Save.**
Add Media Resources to Device Pool

You can make MOH available to devices by assigning the media resource group list that contains the MOH resource to a device or to the device pool.

Note

The device in a call will use the media resource group list that is assigned to the device in the Phone Configuration window. If none is assigned, it will use the media resource group list for the device pool that is used for the call.

Procedure

Step 1
From Cisco Unified CM Administration, do either of the following:
- Choose System > Device Pool.
- Choose Device > Phone.

Step 2
Click Find and select an existing phone or an existing device pool.

Step 3
From the Media Resource Group List drop-down, select the media resource group list that contains the Music On Hold resource.

Step 4
Complete the remaining fields in the configuration window. For help with the fields and their settings, see the online help.

Step 5
Click Save.

Configure MOH Service Parameters

Use this procedure to configure optional service parameters for Music On Hold (MOH). For many deployments the default settings will be sufficient.

Procedure

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

Step 2
From the Server drop-down list, select the server.

Step 3
From the Service drop-down list, select Cisco IP Voice Media Streaming.

Step 4
From the Clusterwide Parameters (Parameters that apply to all servers) area, configure optional MOH service parameters.

Step 5
Click Save.

Step 6
From the Service drop-down list, select Cisco CallManager.

Step 7
Configure optional MOH parameters. For example, under Clusterwide Parameters (Service), you can assign the default audio sources for Hold.

Step 8
Click Save.
Note   All parameters apply only to the current server except parameters that are in the cluster-wide group.

---

**View Music on Hold Audio File**

View existing Music On Hold audio files that are stored on the system.

**Procedure**

**Step 1**  
In Cisco Unified CM Administration, choose Media Resources > MOH Audio File Management. The Music On Hold Audio File Management window appears.

**Step 2**  
View the following information for each record:

- **Check box**—If the audio file can be deleted, a check box appears before the File Name column.
- **File Name**—This column displays the audio file name.
- **Length**—This column displays the audio file length in minutes and seconds.
- **File Status**—This column displays one of the following statuses of an audio file:
  - **Translation Complete**—This status appears after a file is uploaded successfully and is available for use as audio files for a music on hold audio source.
  - **In Use**—This status appears after you add a Music On Hold audio source that uses this audio file as its MOH audio source file.

**Note** You cannot delete a file with In Use status.

---

**Unicast and Multicast Audio Sources**

Unicast Music On Hold is the system default option. However, you need to configure for multicast, if required. Both multicast and unicast configurations present the same audio-source behavior to held parties. Each audio source is used once, and the stream is split internally and is sent to the held parties. The only difference between multicast and unicast, in this case, is how the data is sent over the network.

**Table 56: Differences Between Unicast and Multicast Audio Sources**

<table>
<thead>
<tr>
<th>Unicast Audio Source</th>
<th>Multicast Audio Source</th>
</tr>
</thead>
<tbody>
<tr>
<td>Consists of streams that are sent directly from the MOH server to the endpoint that requests an MOH audio stream.</td>
<td>Consists of streams that are sent from the MOH server to a multicast group IP address. Endpoints that request an MOH audio stream can join multicast MOH, as needed.</td>
</tr>
</tbody>
</table>
### Unicast and Multicast Audio Sources

<table>
<thead>
<tr>
<th><strong>Unicast Audio Source</strong></th>
<th><strong>Multicast Audio Source</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>A unicast MOH stream is a point-to-point, one-way audio RTP stream between the server and the endpoint device.</td>
<td>A multicast MOH stream is a point-to-multipoint, one-way audio RTP stream between the MOH server and the multicast group IP address.</td>
</tr>
<tr>
<td>Unicast MOH uses a separate source stream for each user or connection. As more endpoint devices go on hold through a user or network event, the number of MOH streams increases.</td>
<td>Enables multiple users to use the same audio source stream to provide MOH.</td>
</tr>
<tr>
<td>An MOH audio source may be configured with an initial (greeting) announcement, which will be played to unicast held parties. For unicast MOH users, this announcement is heard from the beginning.</td>
<td>For multicast users, this announcement is not heard.</td>
</tr>
<tr>
<td>The additional MOH streams can have a negative effect on network throughput and bandwidth.</td>
<td>Multicast MOH conserves system resources and bandwidth.</td>
</tr>
<tr>
<td>Extremely useful in networks in which multicast is not enabled or devices are incapable of multicast.</td>
<td>Can be problematic in situations in which a network is not enabled for multicast or the endpoint devices are incapable of processing multicast.</td>
</tr>
<tr>
<td>Includes managing devices only.</td>
<td>Includes managing devices, IP addresses, and ports.</td>
</tr>
<tr>
<td>No requirement to define the Music On Hold server.</td>
<td>Administrators must define at least one audio source to allow multicasting. To define Music On Hold servers for multicast, first define the server to allow multicasting.</td>
</tr>
<tr>
<td>Functions without configuring MOH audio source, MOH server, or media resource group list.</td>
<td>Functions only if both media resource groups and media resource group lists are defined to include a multicast Music On Hold server. For media resource groups, you must include a Music On Hold server that is set up for multicast. These servers are labeled as (MOH) [Multicast]. Also, check the <strong>Use Multicast for MOH Audio</strong> check box when you define a media resource group for multicast.</td>
</tr>
</tbody>
</table>

**Note**

The Multicast MOH Direction Attribute for SIP service parameter determines whether Cisco Unified Communications Manager sets the direction attribute of the Session Description Protocol (SDP) in its multicast Music On Hold (MOH) INVITE message to sendOnly or recvOnly.

If your deployment uses SIP phone uses Release 8.4 and earlier for Cisco Unified IP Phones 7940 and 7960, or SIP phone uses Release 8.1(x) and earlier for Cisco Unified IP Phones 7906, 7911, 7941, and 7961, set this parameter to **sendOnly**. Otherwise, leave this parameter set to the default value, **recvOnly**.
# Music On Hold Interactions and Restrictions

## Music On Hold Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
</table>
| Multicast Music On Hold over H.323 Intercluster Trunks | Using the multicast MOH over H.323 intercluster trunk feature, you can multicast MOH to work over H.323 intercluster trunks (ICT). When a call connects over an intercluster trunk and one of the parties presses the Hold key, MOH streams over the intercluster trunk. If you have turned on the multicast MOH and have configured the holding party and trunk to use the multicast MOH server, MOH streams with multicast. Only one multicast MOH stream streams over the trunk regardless of the number of calls that are put on hold on this trunk. Additional points regarding this feature:  
  - This feature does not work if any middle box between Cisco Unified Communications Managers does not pass the new fields in Terminal Capability Set (TCS) and OLC message.  
  - This feature requires no additional configuration for field up multicast MOH, and applies only between Cisco Unified Communications Managers that support single-transmitter multicast.  
  - The feature is On by default, but can be turned off by setting the `Send Multicast MOH in H.245 OLC Message` service parameter to `False`. Setting this value can resolve interoperability issues that the feature might cause. |
| Music On Hold Failover and Fallback | The MOH server supports Cisco Unified Communications Manager lists and failover as implemented by the software conference bridge and media termination point. Upon failover, the system maintains connections to a backup Cisco Unified Communications Manager, if available. When a Music On Hold server fails during an active Music On Hold session, the held party hears no music from this point. However, this situation does not affect normal call functions. |
| Call Park and Directed Call Park | Music On Hold allows users to place calls on hold with music that a streaming source provides. Music On Hold allows two types of hold:  
  - User hold—The system invokes this type of hold when a user presses the Hold button or Hold softkey.  
  - Network hold—This type of hold takes place when a user activates the Transfer, Conference, or Call Park feature, and the hold automatically gets invoked. This hold type applies to directed call park because directed call park is a transfer function. However, Directed Call Park uses the Cisco Call Manager service parameter, Default Network Hold MOH Audio Source, for the audio source. |
Interaction

Examples include RSVP Agent, TRP, Music On Hold (MOH), MTP, transcoder, and conference bridge. Media resources are local to the visiting phone (other than RSVP Agents).

Hold Reversion
Cisco Unified Communications Manager supports MOH on a reverted call if MOH is configured for a normal held call.

Media Resource Selection
Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a Music On Hold resource.

Secured Music On Hold with SRTP
Cisco Unified Communications Manager enhances the Cisco IP Voice Media Streaming application service to support Secure Real-Time Protocol (SRTP). Hence, when you enable the Cisco Unified Communications Manager cluster or system for security, the MOH server registers with Cisco Unified Communications Manager as an SRTP capable device. If the receiving device is also SRTP-capable, the music media is encrypted before streaming to the receiving device.

Make sure of the following:

- Cluster security should be mixed mode—Run the `utils ct1 set-cluster mixed-mode` CLI command
- SIP trunks in the path support SRTP—The SRTP Allowed check box must be checked in the **Trunk Configuration** window for SRTP to work over the trunk.
- Devices support SRTP—In the Phone Security Profile used by the endpoint, the **Device Security Mode** must be **Encrypted**.

### Music On Hold Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multicast Music On Hold Support</td>
<td>Computer Telephony Integration (CTI) and media termination point (MTP) devices do not support the multicast Music On Hold feature. If you configure CTI or MTP devices with a multicast MoH device in the media resource group list of the CTI device, call control issues may result. CTI and MTP devices do not support multicast media streaming.</td>
</tr>
<tr>
<td>Distribution of fixed-device audio sources</td>
<td>Cisco Unified Communications Manager does not support distribution of fixed-device (hardware) audio sources across Music On Hold servers within a media resource group.</td>
</tr>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Unacceptable Audio Quality with G.729a codec</td>
<td>Because the G.729a codec is designed for human speech, if you use it with Music On Hold for music, it may not provide acceptable audio quality.</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager System Support</td>
<td>A Cisco Unified Communications Manager cluster or system supports only virtualized deployments on Cisco Unified Computing System (UCS) servers or other Cisco-approved third-party server configurations. You cannot use the Music On Hold feature with an external source (USB audio dongle) for the nodes that provide MOH from an external source.</td>
</tr>
<tr>
<td>Multicast Support</td>
<td>The administrator can designate a Music On Hold server as either unicast or multicast, provided that resources exist to support multicast.</td>
</tr>
<tr>
<td>Caller-specific MOH Support</td>
<td>Caller-specific MOH is not supported when calls are received or transferred over QSIG tunneling-enabled SIP trunks.</td>
</tr>
<tr>
<td>MP3 Format Support</td>
<td>The Music On Hold feature does not support the MP3 format.</td>
</tr>
<tr>
<td>Interoperability between H.323 and SIP Protocols</td>
<td>Multicast MOH does not support interoperability between H.323 and SIP protocols.</td>
</tr>
<tr>
<td>SRTP Support</td>
<td>Multicast MoH audio streams are not encrypted and do not support SRTP.</td>
</tr>
<tr>
<td>Multicast Streams</td>
<td>MTPs do not support multicast streams.</td>
</tr>
<tr>
<td>Encryption of Multicast Music On Hold RTP Streams</td>
<td>Cisco Unified Communications Manager does not support encryption of multicast Music On Hold RTP streams. For secure MOH audio, you should not configure multicast audio sources.</td>
</tr>
<tr>
<td>Fixed Music On Hold Device</td>
<td>The fixed Music On Hold device cannot specify an audio source that connects through a USB, because Cisco Unified Communications Manager does not support USB when running on VMware. However, VMware supports internal Music On Hold.</td>
</tr>
<tr>
<td>MOH Server Failure</td>
<td>Cisco Unified Communications Manager takes no action when a Music On Hold server fails during an active Music On Hold session.</td>
</tr>
<tr>
<td>Multicast MOH</td>
<td>When an MTP resource gets invoked in a call leg at a site that is using multicast MOH, Cisco Unified Communications Manager falls back to unicast MOH instead of multicast MOH.</td>
</tr>
<tr>
<td>Provisioning</td>
<td>If you do not provision the user and network MOH audio source identifiers, or if one or both values are invalid, the caller-specific MOH information in the SIP header is ignored. The call reverts to tone on hold and an invalid MOH audio source alarm is raised.</td>
</tr>
<tr>
<td>Header Values</td>
<td>• When both the user and network MOH audio source identifiers are present in the header, any invalid value is replaced by the default value (0).</td>
</tr>
<tr>
<td></td>
<td>• If both values are zero, or the only value is zero, the header in the incoming INVITE is ignored.</td>
</tr>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| MOH Audio Source Identifier                  | • If you provide only one MOH audio source identifier in the SIP header, including if a comma appears before or after the MOH audio source identifier value, the same MOH ID is used for both user and network MOH. The SIP trunk populates both the user and the network MOH audio source identifiers in the SIP header so that Call Control always receive both values.  
  • If there are more than two MOH audio source identifier values separated by a comma in the header, then the first two values are used. Subsequent values are ignored. |
| Administrators for Consistent Caller-specific MOH Configurations | Administrators are responsible to maintain consistent caller-specific MOH configurations when multiple Cisco Unified Communications Manager clusters are involved.                                                      |
| Original Incoming Caller                      | The original incoming caller to the call center cannot change during the course of the entire call.                                                                                                |
| MOH Information                               | The Music On Hold information is shared only across SIP trunks.                                                                                                                                         |

**Music On Hold Troubleshooting**

**Music On Hold Does Not Play on Phone**

Phone user cannot hear Music On Hold.

- G.729a codec is used with MOH for music, which may not provide acceptable audio quality.
- An MTP resource is invoked in a call leg at a site that is using multicast MoH.
- When an MTP resource gets invoked in a call leg at a site that is using multicast MoH, the caller receives silence instead of Music On Hold. To avoid this scenario, configure unicast MoH or Tone on Hold instead of multicast MoH.
Self Care Portal Overview

From the Cisco Unified Communications Self Care Portal, users can customize features and settings for their phones.

As the administrator, you control access to the Self Care Portal. Before an end user can access the portal, you must add the user to the default **Standard CCM End Users** access control group, or to any access control group that has the **Standard CCM End Users** role assignment. In addition, users require their user ID, password, and the URL with which to access the portal. Users can access the portal via one of these URLs:

- `http://<server_name:portnumber>/ucmuser/`, where `server_name` is the host on which the web server is installed and `portnumber` is the port number on that host.
- `http://<ip address>/ucmuser/` or `http://<ip address>/ccmuser/`, where `ip address` is the host on which the web server is installed.

Optionally, you can also configure enterprise parameters within Cisco Unified Communications Manager in order to assign which phone settings are available for end users to configure. For example, the **Show Call Forwarding** enterprise parameter determines whether users can configure Call Forward via the portal.

Self Care Portal Task Flow

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Grant User Access to the Self Care Portal, on page 628</td>
<td>To access the portal, end users must be assigned to the <strong>Standard CCM End Users</strong> access control group or to any group that has the <strong>Standard CCM End Users</strong> role assignment.</td>
</tr>
</tbody>
</table>
Grant User Access to the Self Care Portal

To access the portal, end users must be assigned to the Standard CCM End Users access control group or to any group that has the Standard CCM End Users role assignment.

Procedure

Step 1
From Cisco Unified Communications Manager Administration, select User Management > End User.

Step 2
Search for the user for whom you want to provide Self-Care access.

Step 3
In the End User section, ensure that the user has a password and PIN configured. Usually these credentials are entered when a new user is added.

Step 4
In the Permission Information section, click Add to Access Control Group.

Step 5
Click Find and select the Standard CCM End Users group or a customized group that contains the Standard CCM End Users role.

Note
For information on editing and configuring access control groups, and role assignments for access control groups, refer to the "Configure User Access" chapter of the System Configuration Guide for Cisco Unified Communications Manager.

Step 6
Select Save.

What to do next
Configure the Self Care Portal Options, on page 628

Configure the Self Care Portal Options

Use this procedure to configure Self Care Portal enterprise parameters in order to control what configuration options are available to users whom access the portal.

Before you begin
Grant User Access to the Self Care Portal, on page 628

Procedure

Step 1
From Cisco Unified Communications Manager Administration, select System > Enterprise Parameters.

Step 2
Under Self Care Portal Parameters, set the Self Care Portal Default Server by selecting one of the available servers from the drop-down list.
This parameter determines which Cisco Unified CM server Jabber uses to display embedded Self Care options pages. If you select None, Jabber defaults to the Publisher.

**Step 3** Configure any of the remaining Self Care Portal Parameters to enable or disable features for the portal. For help with the fields, refer to the enterprise parameters help.

**Step 4** Select Save.

---

### Self Care Portal Interactions and Restrictions

The following table highlights feature interactions and restrictions with the Self-Care Portal.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction or Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Authenticated user https request</td>
<td>When an authenticated user makes a request to https://{CUCM_address}/ucmuser/hostAlive/{host}, the following happens:</td>
</tr>
<tr>
<td></td>
<td>• If the request is successful at getting http://{host}/ or if the request can ping {host} Cisco Unified Communications Manager returns the string, &quot;true&quot;.</td>
</tr>
<tr>
<td></td>
<td>• If the request is unsuccessful, Cisco Unified Communications Manager returns the string &quot;false&quot;.</td>
</tr>
<tr>
<td>Maximum Login for Extension Mobility</td>
<td>For end users to be able to configure this setting within the Self-Care Portal, an administrator must have checked the <strong>Allow End User to set their Extension Mobility maximum login time</strong> option in the associated User Profile of Cisco Unified CM Administration.</td>
</tr>
<tr>
<td></td>
<td>If this option is selected within the User Profile, for all users whom use the profile, the Self-Care Portal setting overrides the administrator-configured values of the <strong>Intra-cluster Maximum Login Time</strong> and <strong>Inter-cluster and Maximum Login Time</strong> service parameters in Cisco Unified Communications Manager.</td>
</tr>
</tbody>
</table>
Emergency Call Handler Overview

Emergency Call Handler helps you to manage emergency calls in your telephony network while following local ordinances and regulations.

When an emergency call is made the following is required:

- The emergency call must be routed to the local Public-Safety Answering Point (PSAP) based on the location of the caller.
- The caller’s location information must be displayed at the emergency operator terminal. The location information can be obtained from an Automatic Location Information (ALI) database.

The caller’s location is determined by the Emergency Location Identification Number (ELIN). An ELIN is a Direct Inward Dial (DID) number that the PSAP can dial to reconnect to the emergency caller if the emergency call is cut off or if the PSAP needs to talk to the caller again. The emergency call is routed to the PSAP based on the location information that is associated with this number.

For multiline phone systems, such as an office system, the ELIN can be associated with more than one telephone by grouping the phones in an ELIN group. An ELIN group in Emergency Call Handler identifies a location. The ELINs under this ELIN group must be mapped to the location in the ALI database.

Each location should have as many ELINs created as needed to support simultaneous emergency calls. For example, to support five simultaneous calls, five ELINs would be needed in an ELIN group.

Note

Emergency Call Handler supports a maximum of 100 ELINs per cluster.

The following types of phone are supported to use ELIN groups:

- SIP and SCCP IP phones
Emergency Call Handler Prerequisites

Example

Before deploying Emergency Call Handler in your network, we recommend that you test the ALI submission process. With your service provider’s help, test that the PSAP can successfully callback into your network using the ALI data.

Reserve the ELIN number from your local PSAP. Ordinances and regulations can differ across different locations and across different companies, so research your security and legal needs before deploying this feature.

Emergency Call Handler Task Flow

Before you begin

• Review Emergency Call Handler Prerequisites, on page 632

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Enable the Emergency Call Handler feature on Cisco Unified Communications Manager. Emergency Call Handler provides essential emergency call features and supports a limited number of locations with phone location assignment by static configuration. If you require advanced emergency call features, such as a greater amount of specific locations or dynamic location assignment, consider Cisco Emergency Responder.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure an Emergency Location (ELIN) Group for a particular site or location.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure device pools to use an Emergency Location (ELIN) Group.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure a particular device to use a particular Emergency Location (ELIN) Group. If you want to use the device pool ELIN Group that is</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td></td>
<td>associated for this device, you can ignore this section.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> Configurations that are made at the device level will overwrite any configurations that were made at the device pool level.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>Enable Route Patterns and Translation Patterns, on page 636</strong></td>
</tr>
<tr>
<td></td>
<td>Enable the Emergency Location (ELIN) service for a route pattern or a translation pattern.</td>
</tr>
<tr>
<td></td>
<td><strong>Caution</strong> No Calling Party Transformation masks are set at the Gateway or Trunk, because these may transform the ELIN that is set by Emergency Call Handler.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> It is mandatory that you enable either route patterns or translation patterns, but it is possible to enable both.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>(Optional) Use the following procedures to perform bulk administration tasks on ELIN group information and phones:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Import Emergency Location Group Information, on page 637</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Export Emergency Location Group Information, on page 638</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Update Phones with a new Emergency Location Group, on page 638</strong></td>
</tr>
<tr>
<td></td>
<td>This section provides information about the Bulk Administration tasks you can use to update ELIN group information and to add phones to new ELIN groups. For Bulk Administration, see the <em>Cisco Unified Communications Manager Bulk Administration Guide, Release 11.0(1)</em>.</td>
</tr>
</tbody>
</table>

**Enable Emergency Call Handler**

Enable the Emergency Call Handler feature on Cisco Unified Communications Manager. Emergency Call Handler provides essential emergency call features and supports a limited number of locations with phone location assignment by static configuration. If you require advanced emergency call features, such as a greater amount of specific locations or dynamic location assignment, consider Cisco Emergency Responder.

**Note**

Do not enable this feature if you are already using an external emergency calling solution such as Cisco Emergency Responder.

If you decide to enable this feature, make sure you disable the external one.
## Configure Emergency Location Groups

Configure an Emergency Location (ELIN) Group for a particular site or location.

### Before you begin

**Enable Emergency Call Handler, on page 633**

### Procedure

**Step 1** From Cisco Unified CM Administration, choose **Call Routing > Emergency Call Handler > Emergency Location Configuration**.

**Step 2** From the Emergency Location Configuration window:

- To enable the Emergency Call Handler feature, check the **Enable Emergency Location (ELIN) Support** check box. The setting default is Disabled. When enabled, the settings related to this feature appear in the Related Settings pane. You must configure these settings for the feature to work. Refer to the tasks below for further details on how to configure these related settings.

- To disable the Emergency Call Handler feature, uncheck the **Enable Emergency Location (ELIN) Support** check box.

**Note** If you disable this feature, all related settings that are configured will be removed. See the Related Settings Pane for all configured settings.

**Note** If you want to disable the feature and you have more than 500 devices associated with ELIN Groups, then you must manually delete the associations until there are fewer than 500 associations before you can disable the feature.

**Step 3** Click **Save**.

### What to do next

Configure Emergency Location Groups, on page 634
Add a Device Pool to an Emergency Location Group

Configure device pools to use an Emergency Location (ELIN) Group.

**Before you begin**
Configure Emergency Location Groups, on page 634

**Procedure**

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

**Step 2**
In the **Find and List Device Pools** window, if you are adding an existing device pool, click **Find** and choose the device pool from the list. If you are adding a new device pool click **Add New**.

**Step 3**
In the **Device Pool Configuration** window, choose the ELIN group to which you want to add the device pool from the **Emergency Location (ELIN) Group** drop-down list. If you are adding a new device pool, fill out any other required fields.

**Step 4**
Click **Save**.

**Note**
Configurations that are made at the device level will overwrite any configurations that were made at the device pool level.

**Note**
The devices that you add to the ELIN Group, should be added to the ELIN Group that represents the particular location at which those devices are located.

**Before you begin**
Add a Device Pool to an Emergency Location Group, on page 635

Add Device to an Emergency Location Group

Configure a particular device to use a particular Emergency Location (ELIN) Group. If you want to use the device pool ELIN Group that is associated for this device, you can ignore this section.
Enable Route Patterns and Translation Patterns

Enable Route Patterns and Translation Patterns

Enable the Emergency Location (ELIN) service for a route pattern or a translation pattern.

**Note**
It is mandatory that you enable either route patterns or translation patterns, but it is possible to enable both.

**Before you begin**
Add Device to an Emergency Location Group, on page 635

**Procedure**

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

**Note**
To enable a translation pattern, choose Call Routing > Translation Pattern.

**Step 2**
In the Find and List Route Patterns or Find and List Translation Patterns window, click Find and choose a route pattern or translation pattern from the list.

**Step 3**
In the Route Pattern Configuration or Translation Pattern Configuration window, check the Is an Emergency Services Number check box.

**Note**
Check this check box only if you are using Emergency Call Handler and not another external emergency calling solution such as Cisco Emergency Responder.
Step 4  Click Save.

### Bulk Administration of Emergency Location Groups and Phones

- **Bulk Administration of Emergency Location Groups and Phones Task Flow, on page 637**

### Bulk Administration of Emergency Location Groups and Phones Task Flow

This section provides information about the Bulk Administration tasks you can use to update ELIN group information and to add phones to new ELIN groups. For more information about Bulk Administration, see the *Cisco Unified Communications Manager Bulk Administration Guide, Release 11.0(1)*.

**Note**

Before you perform these procedures, make sure that you have enable the Emergency Call Handler feature. See *Enable Emergency Call Handler, on page 633*.

<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>Import Emergency Location Group Information, on page 637</td>
<td>Import Emergency Location (ELIN) Group information using the Bulk Administration Tool.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Export Emergency Location Group Information, on page 638</td>
<td>Export Emergency Location (ELIN) Group information using the Bulk Administration Tool.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Update Phones with a new Emergency Location Group, on page 638</td>
<td>Find and list multiple phones and configure them with a new Emergency Location (ELIN) Group.</td>
</tr>
</tbody>
</table>

**Import Emergency Location Group Information**

Import Emergency Location (ELIN) Group information using the Bulk Administration Tool.

**Procedure**

- **Step 1**: From Cisco Unified CM Administration, choose *Bulk Administration > Import/Export > Import*.
- **Step 2**: From the *File Name* drop-down list, choose the name of the .tar file you want to import, and click *Next*.
- **Step 3**: The *Import Configuration* section lists all the components of the .tar file. Check the ELIN Group-related check boxes for the options that you want to import.
- **Step 4**: Choose to run the job immediately or later by clicking the corresponding radio button.
- **Step 5**: To create a job for importing the selected data, click *Submit*. A message in the Status section notifies you that the job was submitted successfully.
- **Step 6**: Use the Job Scheduler option in the Bulk Administration main menu to schedule or activate this job.
What to do next
Export Emergency Location Group Information, on page 638

Export Emergency Location Group Information

Export Emergency Location (ELIN) Group information using the Bulk Administration Tool.

Before you begin
Import Emergency Location Group Information, on page 637

Procedure

Step 1 From Cisco Unified CM Administration, choose Bulk Administration > Import/Export > Export.
Step 2 In the Export Data window, in the Job Information pane, enter the .tar file name, without the extension, in the Tar FileName field. BPS uses this filename to export the configuration details.

Note All files that are exported at the same time get bundled together (.tar) and can be downloaded from the server.

Step 3 To export ELIN Group information, check the Elin Group check box on the Select items to Export pane.
Step 4 (Optional) Perform these steps:
  • To export device pools with ELIN Groups configured, check the Device Pools check box.
  • To export phones with ELIN Groups configured, check the Phone check box.

Step 5 In the Job Description field, enter the description that you want to override for the job. Export Configuration is the default description.
Step 6 You can choose to run the job immediately or later by clicking the corresponding radio button.
Step 7 To create a job for exporting the selected data, click Submit. A message in the Status pane notifies you that the job was submitted successfully.
Step 8 Use the Job Scheduler option in the Bulk Administration main menu to schedule or activate this job.

What to do next
Update Phones with a new Emergency Location Group, on page 638

Update Phones with a new Emergency Location Group

Find and list multiple phones and configure them with a new Emergency Location (ELIN) Group.

Before you begin
Export Emergency Location Group Information, on page 638

Procedure

Step 1 From Cisco Unified CM Administration, choose Bulk Administration > Phones > Update Phone > Query.
Step 2 In the **Find and List Phones To Update** window, set the parameters for your search and click **Find**.

**Note** To update all phones, click **Find** and do not specify a query.

Step 3 The **Find and List Phones To Update** window displays the details of the phones that you chose. Click **Next**.

Step 4 In the **Update Phones** window, check the **Emergency Location (ELIN) Group** check box, and choose a new ELIN Group from the drop-down list.

Step 5 Click **Submit**.

---

### Emergency Call Handler Interactions and Restrictions

- Interactions, on page 639
- Restrictions: None

#### Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb Call</td>
<td>Calls made by PSAP CallBack will overwrite a Do Not Disturb (DND) configuration of a destination device.</td>
</tr>
<tr>
<td>Reject</td>
<td>If DND Call Reject is enabled, when the emergency number is dialed using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected and the ELIN is called back using PSAP CallBack, the call is routed to the phone irrespective of the phone's DND settings.</td>
</tr>
<tr>
<td>Call Forward All</td>
<td>Calls made by PSAP CallBack will overwrite Call Forward All (CFA) settings of the destination device.</td>
</tr>
<tr>
<td></td>
<td>If a phone has CFA enabled and if the emergency number using the translation pattern is dialed, an ELIN will be associated for this outbound emergency call. If the call is disconnected and the ELIN is called back using PSAP CallBack, the call is routed to the phone irrespective of the phone's CFA settings.</td>
</tr>
<tr>
<td>Single Number Reach</td>
<td>PSAP CallBack will ignore the Single Number Reach (SNR) configuration.</td>
</tr>
<tr>
<td></td>
<td>When a phone has SNR enabled with the Remote Destination pointing to a mobile number. If the emergency number is dialed using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected, and the ELIN number is called back using PSAP CallBack, the call is routed to the phone and not to the remote destination.</td>
</tr>
<tr>
<td>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Extension Mobility</td>
<td>PSAP CallBack call will consider Extension Mobility (EM) status. If you log in with EM profile credentials and dial the emergency number using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected and the ELIN where the user is still logged in is called back using PSAP CallBack, the call is routed to the device which initiated the call. <strong>Note</strong>: This is the device on which the user is still logged in.</td>
</tr>
<tr>
<td></td>
<td>PSAP CallBack will fail if a user logs out of EM before a PSAP CallBack is performed. When a user logs in with EM profile credentials, and the emergency number is dialed using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected and is called back using PSAP CallBack, if the user has since logged out, the call will not route to the device that initiated the call and will fail.</td>
</tr>
<tr>
<td></td>
<td>PSAP CallBack with a user logged in on a different device. When a user logs in with EM profile credentials at Phone A and dials the emergency number using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected, the user should log out from Phone A. If the user then logs in to another phone, Phone B, with the same profile, and the ELIN is called back using PSAP CallBack, the call is then be routed to Phone B with normal priority, meaning CFA settings will be ignored and DND settings will not be ignored.</td>
</tr>
<tr>
<td></td>
<td>PSAP CallBack call with multiple logins. When a user logs in with EM profile credentials at Phone A and dials the emergency number using the translation pattern, an ELIN number will be associated for this outbound emergency call. If the call is disconnected and the user logs in to another phone, Phone B, with the same profile while the user is still logged in on Phone A, and the ELIN is called back using PSAP CallBack, then the call is routed to Phone A only, the device on which the call originated.</td>
</tr>
<tr>
<td>Device Mobility</td>
<td>A roaming device will use the Roaming Device Pool's ELIN Group for an outbound emergency call. Move a device with Device Mobility enabled from its home location to the Roaming location, a change in IP subnet, so that it gets associated with the Roaming device pool. If the emergency number is dialed using the translation pattern, an ELIN is associated for this outbound emergency call. The ELIN belongs to the ELIN Group that is associated with the Roaming Device Pool.</td>
</tr>
</tbody>
</table>
Shared Lines

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>PSAP CallBack</td>
<td>rings only on the device which made the emergency call even if the line is shared by different devices.</td>
</tr>
<tr>
<td></td>
<td>Phone A and Phone B share a Directory Number (DN). If the emergency number is dialed using the translation pattern, an ELIN is associated for this outbound emergency call. If the call is disconnected, and the ELIN is called back using PSAP CallBack, the call is routed to Phone A only, the device from which the call originated.</td>
</tr>
</tbody>
</table>

## Emergency Call Handler Troubleshooting

### About Emergency Call Handler Troubleshooting Scenarios

This section provides information about some Emergency Call Handler troubleshooting scenarios in the following areas:

- Configuration Scenarios
- Outgoing Calls Scenarios
- Incoming Calls Scenarios

### Configuration Scenarios

#### Emergency Calls Get Busy Signals and Are Not Routed

**Problem:**

Emergency calls get busy signals and are not routed.

**Solution:**

If a user who is dialing the emergency call is running a reorder tone, perform the following checks:

- Check whether the translation or route pattern for the emergency call has been used. This may require checking for the device or phone on CSS.

- Check whether the **Is an Emergency Services Number** check box has been checked for the translation or route pattern of the emergency call, and that it is correctly routing to the gateway.

If the user who is dialing the emergency call is not reaching the correct gateway or Public Service Answering Point (PSAP), check that the settings or device pool settings for the phone or device are configured with the correct Emergency Location (ELIN) Group.

#### Emergency Location Numbers Are Dialed from Outside Running a Reorder Tone

**Problem:**

Emergency Location (ELIN) numbers are dialed from outside while running a reorder tone.
Cause:
In this case the ELINs have been set as DID which is used to identify a caller's location. This should not be used on any phone or for any other purpose.

Solution:
Check the ELIN configuration information, and unset the ELINs that have been set as DID.

Outgoing Calls Scenarios

Outgoing Emergency Call Does Not Contain Calling Party as Emergency Location Number

Problem:
An outgoing emergency call does not contain the calling party as an Emergency Location (ELIN) number.

Cause:
The translation pattern or route pattern for this ELIN was not configured correctly.

Solution:
Check that the translation pattern or route pattern settings are correctly configured for this ELIN, and make sure that the Is an Emergency Services number check box is checked on the relevant translation pattern or route pattern configuration page.

Outgoing Emergency Call Contains Modified Emergency Location Number

Problem:
An outgoing emergency call contains a modified Emergency Location (ELIN) number.

Cause:
The outgoing trunk or route list contains extra transformations that are not required for ELINs.

Solution:
Check the transformations that were applied for the call, and make sure that only the required transformations for ELINs are present on the outgoing trunk or route list.

Incoming Calls Scenarios

Incoming PSAP Callback Call Fails

Problem:
An incoming PSAP Callback call fails.

Cause:
The device that made the original emergency call was not registered correctly.

Solution:
Check whether the device that made the original emergency call is still registered and whether any Extension Mobility is involved.
Incoming PSAP CallBack Call is Not Routed as Expected

Problem:
An incoming PSAP CallBack call does not get routed as expected.

Cause:
The Emergency Location (ELIN) number does not match the number of the original dialed party.

Solution:
For an ELIN to be successfully reverse mapped to the original dialed party, these two numbers must match. If there are already transformations at the incoming Gateway or Trunk and significant digits configured, make sure that the final transformed called party matches the ELIN number.
Incoming PSAP CallBack Call is Not Routed as Expected
Enterprise Groups Overview

When Enterprise Groups is configured, Cisco Unified Communications Manager includes user groups when it synchronizes its database with an external LDAP directory. In Cisco Unified CM Administration, you can view synced groups in the User Groups window.

This feature also helps administrators to:

- Provision users with similar characteristics traits with a comment set of features (for example, the sales and accounting teams).
- Target messages to all users in a specific group.
- Configure uniform access for all members of a specific group.

This feature also helps Cisco Jabber users to quickly build contact lists of users who shares common traits. Cisco Jabber users can search the external LDAP Directory for user groups and then add them to their contact list. For example, a Jabber user can search the external LDAP directory and add the sales group to a contact list, thereby adding all of the sales team members into the contact list as well. If the group gets updated in the external directory, the user's contact list is updated automatically.

Enterprise Groups is supported with Microsoft Active Directory on Windows as the external LDAP directory.

Note

If you disable the Enterprise Groups feature, Cisco Jabber users cannot search for enterprise groups or see the groups that they already added to their contact lists. If a user is already logged in when you disable the feature, the group will be visible until the user logs out. When the user logs in again, the group will not be visible.
Security Groups

Security Groups are a subfeature of Enterprise Groups. Cisco Jabber users can also search for, and add, security groups to their contact list. To set up this feature, administrators must configure a customized LDAP filter and apply it to the configured LDAP directory sync. Security Groups are supported with Microsoft Active Directory only.

Maximum Allowed Entries

When configuring Enterprise Groups, make sure that you configure contact list maximums that handle groups

- The maximum number of entries that are allowed in a contact list is the sum of the number of entries in the contact list and the number of entries in groups that are already added to the contact list.
- Maximum entries in contact list = (number of entries in contact list) + (number of entries in groups)
- When the Enterprise Groups feature is enabled, Cisco Jabber users can add the groups to the contact list if the number of entries in the contact list is less than the maximum allowed entries. If the maximum allowed entries is exceeded while the feature is disabled, the users are not restricted until the feature is enabled. If the user continues to be logged in after the feature is enabled, no error message is displayed. When the user logs out and logs in again, an error message is displayed that asks the users to clear the excess entries.

Enterprise Groups Prerequisites

This feature assumes that you already have an LDAP Directory sync schedule configured with the below conditions. For details on how to configure an LDAP Directory sync, see the "Import Users from LDAP Directory" chapter of the System Configuration Guide for Cisco Unified Communications Manager.

- The Cisco DirSync service must be activated
- The LDAP Directory sync must include both users and groups
- Regular LDAP Directory syncs, as configured with the LDAP Directory Synchronization Schedule must be scheduled.

Supported LDAP Directories

Only Microsoft Active Directory is supported with enterprise groups.

Enterprise Groups Configuration Task Flow

Complete these tasks to configure the Enterprise Groups feature.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Verify Group Sync from LDAP Directory, on page 647</td>
<td>Confirm that your LDAP Directory sync includes both users and groups.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-----------------------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Step 2 Enable Enterprise Groups, on page 647</td>
<td>Complete this task to enable Cisco Jabber users to search for enterprise groups in Microsoft Active Directory and add them to their contact lists.</td>
</tr>
<tr>
<td>Step 3 Enable Security Groups, on page 648</td>
<td>(Optional) If you want Cisco Jabber users to be able to search for and add security groups to their contact lists, complete this task flow.</td>
</tr>
<tr>
<td>Step 4 View User Groups, on page 650</td>
<td>(Optional) View enterprise groups and security groups that are synchronized with Cisco Unified Communications Manager database.</td>
</tr>
</tbody>
</table>

### Verify Group Sync from LDAP Directory

Use this procedure to confirm that your LDAP Directory sync includes users and groups.

#### Procedure

1. From Cisco Unified CM Administration, choose **Server > LDAP > LDAP Directory**.
2. Click **Find** and select the LDAP directory from which you are syncing enterprise groups.
3. Confirm that the **Synchronize** field has **Users and Groups** selected.
4. Complete any remaining fields in the LDAP Directory configuration window. For help with the fields and their settings, refer to the online help.
5. Click **Save**.

#### What to do next

**Enable Enterprise Groups, on page 647**

**Enable Enterprise Groups**

Configure the system to include enterprise groups in LDAP Directory syncs.
Enable Security Groups

If you want to allow Cisco Jabber users to be able to add a security group to their contact list, complete these optional tasks to include security groups in an LDAP Directory sync.

**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 Security Group Filter, on page 649</td>
<td>Create an LDAP filter that filters both directory groups and security groups.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Cisco Jabber for Security Groups, on page 650</td>
<td>Update existing service profiles to give Cisco Jabber users whom are associated to that service profile access to search and add security groups.</td>
</tr>
</tbody>
</table>

**What to do next**

Enable Security Groups, on page 648

---

Enable Security Groups

If you want to allow Cisco Jabber users to be able to add a security group to their contact list, complete these optional tasks to include security groups in an LDAP Directory sync.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Create Security Group Filter, on page 649</td>
<td>Create an LDAP filter that filters both directory groups and security groups.</td>
</tr>
<tr>
<td>Configure Cisco Jabber for Security Groups, on page 650</td>
<td>Update existing service profiles to give Cisco Jabber users whom are associated to that service profile access to search and add security groups.</td>
</tr>
</tbody>
</table>

---

**Note**

Security group sync is supported from Microsoft Active Directory only.

---

**Note**

You cannot add new configurations into an existing LDAP Directory configuration in Cisco Unified Communications Manager where the initial sync has already occurred.

---

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Create Security Group Filter, on page 649</td>
<td>Create an LDAP filter that filters both directory groups and security groups.</td>
</tr>
<tr>
<td>Configure Cisco Jabber for Security Groups, on page 650</td>
<td>Update existing service profiles to give Cisco Jabber users whom are associated to that service profile access to search and add security groups.</td>
</tr>
</tbody>
</table>
Create Security Group Filter

Create an LDAP filter that filters security groups.

Procedure

**Step 1**  
In Cisco Unified CM Administration, choose System > LDAP > LDAP Filter.

**Step 2**  
Click Add New.

**Step 3**  
Enter a unique Filter Name. For example, syncSecurityGroups.

**Step 4**  
Enter the following Filter: (&(objectClass=group)(CN=*)).

**Step 5**  
Click Save.

What to do next

Synchronize Security Groups from LDAP Directory, on page 649

Synchronize Security Groups from LDAP Directory

Add your Security Group filter to an LDAP Directory sync and complete a sync.

Note

You cannot add new configurations into an existing LDAP Directory configuration in Cisco Unified Communications Manager if the initial LDAP sync has already occurred.

Note

For detailed information on how to set up a new LDAP Directory sync, see the "Configure End Users" part of the System Configuration Guide for Cisco Unified Communications Manager.

Before you begin

Create Security Group Filter, on page 649

Procedure

**Step 1**  
In Cisco Unified CM Administration, choose System > LDAP > LDAP Directory.

**Step 2**  
Do one of the following:

- Click Add New to create a new LDAP Directory.
- Click Find and select the LDAP Directory from which the security groups will be synchronized.

**Step 3**  
From the LDAP Custom Filter for Groups drop-down list box, select the security group filter that you created.

**Step 4**  
Click Save.
Configure Cisco Jabber for Security Groups

Update existing service profiles to allow Cisco Jabber users whom are associated to that service profile to add security groups from an LDAP directory to their contact lists.

Note
For information on how to set up new service profiles and assign them to Cisco Jabber users, see the "Configure Service Profiles" chapter of the System Configuration Guide for Cisco Unified Communications Manager.

Before you begin
Synchronize Security Groups from LDAP Directory, on page 649

Procedure

Step 1 Complete any remaining fields in the Service Profile Configuration window. For help with the fields and their settings, refer to the online help.

Step 2 Click Find and select the service profile that your Jabber users use.

Step 3 Under Directory Profile, check the Allow Jabber to Search and Add Security Groups check box.

Step 4 Click Save.
Cisco Jabber users who are associated to this service profile can now search and add security groups.

Step 5 Repeat this procedure for all service profiles that your Cisco Jabber users use.

View User Groups

You can view the enterprise groups and security groups that are synchronized with the Cisco Unified Communications Manager database using the following steps.

Procedure

Step 1 From Cisco Unified CM Administration, choose User Management > User Settings > User Group. The Find and List User Groups window appears.

Step 2 Enter search criteria and click Find.
A list of user groups that match the search criteria is displayed.
**Step 3**
To view a list of users that belong to a user group, click on the required user group. The *User Group Configuration* window appears.

**Step 4**
Enter search criteria and click **Find**. A list of users that match the search criteria is displayed. If you click on a user in the list, the *End User Configuration* window appears.

---

## Enterprise Groups Deployment Models (Active Directory)

The Enterprise Groups feature offers two deployment options for Active Directory.

---

**Important**

Ensure that Cluster 1 and Cluster 2 have a unique set of UserGroup, UserGroupMember, and UserGroupWatcherList records before synchronizing data through the Cisco Intercluster Sync Agent service. If both the clusters have unique sets of records, both the clusters will have a super set of all the records after synchronization.

### Enterprise Groups Deployment Model 1

In this deployment model, Cluster 1 and Cluster 2 synchronize different subsets of users and groups from Microsoft Active Directory. The Cisco Intercluster Sync Agent service replicates the data from Cluster 2 into Cluster 1 to build the complete database of users and groups.

*Figure 12: Enterprise Groups Deployment Model 1*
Enterprise Groups Deployment Model 2

In this deployment model, Cluster 1 synchronizes all the users and groups from Microsoft Active Directory. Cluster 2 synchronizes only users from Microsoft Active Directory. The Cisco Intercluster Sync Agent service replicates groups information from Cluster 1 into Cluster 2.

Caution

If you are using this deployment model, ensure that you synchronize the groups data in only one cluster. The Enterprise Groups feature will not work as expected if you fail to do so.

You can verify your configuration on the Cisco Unified CM IM and Presence Administration > Presence > Inter-Clustering window.

Check the status of the Enterprise Groups LDAP Configuration parameter in the Inter-cluster peer table. No conflict found means there are no misconfigurations between peers. If there are conflicts found, click the Enterprise GroupConflicts link, and click the details button which appears. This opens a Reporting window for a detailed report.

Figure 13: Enterprise Groups Deployment Model 2
## Enterprise Groups Limitations

### Table 57: Enterprise Groups Limitations

<table>
<thead>
<tr>
<th>Limitation</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Block Everyone</td>
<td>When a Cisco Jabber user enables the &quot;Block Everyone&quot; feature from within their Cisco Jabber policy settings, the block prevents other Jabber users from viewing or exchanging IMs and Presence with the blocking user, unless they are listed as a contact in the blocking user's contact list.</td>
</tr>
<tr>
<td></td>
<td>For example, a Cisco Jabber user (Andy) has enabled Block everyone within his personal Jabber settings. The following list breaks down how Andy's block affects other Jabber users whom may or may not be included in Andy's personal contact list. In addition to the block, Andy has a personal contact list that:</td>
</tr>
<tr>
<td></td>
<td>• Includes Bob—Because Bob is in Andy's personal contact list, he can still send IMs and view Andy's presence despite the block.</td>
</tr>
<tr>
<td></td>
<td>• Omits Carol—Carol cannot view Andy's presence or send IMs due to the block.</td>
</tr>
<tr>
<td></td>
<td>• Omits Deborah as a personal contact. However, Deborah is a member of an enterprise group that Andy has listed as a contact—Deborah is blocked from viewing Andy's presence or sending IMs to Andy.</td>
</tr>
<tr>
<td></td>
<td>Note that Deborah is blocked from viewing Andy's presence, or sending IMs to Andy, despite the fact that she is a member of an enterprise group in Andy's contact list. For additional details on enterprise group contacts behavior, see CSCvg48001.</td>
</tr>
<tr>
<td>Intercluster peering with a 10.x cluster</td>
<td>Enterprise Groups is supported for releases 11.0(1) and higher. If the synced group includes group members from a 10.x intercluster peer, users on the higher cluster cannot view the presence of synced members from the 10.x cluster. This is due to database updates that were introduced in 11.0(1) for the Enterprise Groups sync. These updates are not a part of the 10.x releases.</td>
</tr>
<tr>
<td></td>
<td>To guarantee that users homed on the higher cluster can view the presence of group members homed on the 10.x cluster, users on the higher cluster should manually add the 10.x users to their contact lists. There are no presence issues for manually added users.</td>
</tr>
<tr>
<td>Multilevel grouping</td>
<td>Multilevel grouping is not allowed for the group sync.</td>
</tr>
<tr>
<td>Group-only synchronization</td>
<td>When a user group and users are present in the same search base, group-only synchronization is not allowed. Instead, the user group as well as the users are synchronized.</td>
</tr>
<tr>
<td>Limitation</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------------------------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Maximum number of user groups</td>
<td>You can synchronize a maximum of 15000 user groups from Microsoft Active Directory server to the Unified Communications Manager database. Each user group can contain from 1 to 200 users. You can configure the exact amount on the <strong>Cisco Unified CM IM and Presence Administration &gt; System &gt; Service Parameters</strong> window. The maximum number of user accounts in the database cannot exceed 160,000.</td>
</tr>
<tr>
<td>User group migration</td>
<td>If a user group is moved from one organization unit to another, you must perform a full sync on the original unit followed by a full sync on the new unit.</td>
</tr>
<tr>
<td>Local groups</td>
<td>Local groups are not supported. Only groups synchronized from Microsoft Active Directory are supported.</td>
</tr>
<tr>
<td>Group members not assigned to IM and Presence Service nodes</td>
<td>Group members that are not assigned to IM and Presence Service nodes display in the contact list with the presence bubble greyed out. However, these members are considered when calculating a maximum numbers of users allowed in the contact list.</td>
</tr>
<tr>
<td>Migration from Microsoft Office Communication Server</td>
<td>During migration from Microsoft Office Communication Server, the Enterprise Groups feature is not supported until users are fully migrated to the IM and Presence Service node.</td>
</tr>
<tr>
<td>LDAP synchronization</td>
<td>If you change the synchronization option in the <strong>LDAP Directory Configuration window</strong> while the synchronization is in progress, the existing synchronization remains unaffected. For example, if you change the synchronization option from <strong>Users and Groups to Users Only</strong> when the synchronization is in progress, the users and groups synchronization still continues.</td>
</tr>
<tr>
<td>Group search functionality over the Edge</td>
<td>Group search functionality over the Edge is offered in this release, but has not been fully tested. As a result, full support for group searches over the Edge cannot be guaranteed. Full support is expected to be offered in a future release.</td>
</tr>
<tr>
<td>Cisco Intercluster Sync Agent service periodic synchronization</td>
<td>If a group name or a group member name is updated in the external LDAP directory, it gets updated on the Cisco Jabber contact list only after the periodic Cisco Intercluster Sync Agent service synchronization. Typically, the Cisco Intercluster Sync Agent service synchronization occurs every 30 minutes.</td>
</tr>
</tbody>
</table>
**Limitation**

Synchronization of users and user groups through different synchronization agreements in LDAP configuration

**Description**

If users and user groups are synchronized into the Cisco Unified Communications Manager database as part of the same synchronization agreement, the user and group association gets updated as expected in Cisco Unified Communications Manager database after synchronization. However, if a user and user group are synchronized as part of different synchronization agreements, the user and the group may not get associated in the database after the first synchronization. The user and group association in the database depends on the sequence in which the synchronization agreements are processed. If the users are synchronized ahead of the groups, then the groups may not be available in the database for association. In such cases, you must ensure that the synchronization agreement with groups is scheduled ahead of the synchronization agreement with the users. Otherwise, after the groups synchronize into the database, the users will get associated with the groups after the next manual or periodic sync.

---

**Tested Scenario**

In a Intercluster deployment with two clusters Cluster A and Cluster B:

Cluster A has 15K OVA and 15K users enabled for IM and Presence Service out of 160K users that are synced from Active Directory. The tested and supported average number of enterprise groups per user on 15K OVA cluster is 13 enterprise groups.

Cluster B has 25K OVA and 25K users enabled for IM and Presence Service out of 160K users that are synced from Active Directory. The tested and supported average number of enterprise groups per user on 25K OVA is 8 enterprise groups.

The tested and supported sum of user's personal contacts in roster and the contacts from enterprise groups that are in a user's roster is less than or equal to 200.

**Note** In environments with more than 2 clusters these numbers are not supported.
PART XII

Device Management

- Headset Management, on page 659
- Video Endpoints Management, on page 675
Headset Management Overview

This chapter assumes that you have already deployed the Cisco Headset 500 Series to your users. For more information on headset deployment, see the "Cisco Headset 500 Series" chapter in the Cisco Headset 500 Series Administration Guide.

In Cisco Unified CM Administration, you can:

- Remotely configure headset settings such as wireless power range, audio bandwidth, Bluetooth on/off, and more.
- Define and control the firmware running on the headset.
- Get a detailed inventory of all the headsets in your deployment.
- Diagnose and troubleshoot headsets using Remote PRT, headset metrics in Call Management Records (CMR) and alarms.

Headset Management Feature Compatibility

Cisco Headset Management is supported in Unified Communications Manager from 12.5(1) SU1 release onwards with the compatible versions of Cisco Headset, Cisco Jabber, and Cisco Expressway releases. The following table lists the available headset management features depending on the headset, phone, and Unified Communications Manager versions you use.
### Table 58: Headset Serviceability Features for Cisco IP Phones

<table>
<thead>
<tr>
<th>New Serviceability Feature</th>
<th>Unified CM 12.5(1) or earlier + Phone Firmware 12.1(1) or earlier</th>
<th>Unified CM 12.5(1)SU1 or earlier + Phone Firmware 12.1(1) or earlier</th>
<th>Unified CM 12.5(1) or earlier + Phone Firmware 12.5(1)</th>
<th>Unified CM 12.5(1)SU1 or earlier + Phone Firmware 12.5(1)</th>
<th>Unified CM 12.5(1) or earlier + Phone Firmware 12.5(1)SR3</th>
<th>Unified CM 12.5(1)SU1 or earlier + Phone Firmware 12.5(1)SR3</th>
</tr>
</thead>
<tbody>
<tr>
<td>COP file installation</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>—</td>
</tr>
<tr>
<td>Manual remote configuration</td>
<td>—</td>
<td>—</td>
<td>X</td>
<td>N/A</td>
<td>X</td>
<td>—</td>
</tr>
<tr>
<td>Headset firmware management on Unified CM</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X</td>
</tr>
<tr>
<td>Remote headset configuration through Unified CM</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X</td>
</tr>
<tr>
<td>Headset inventory on Unified CM</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X*</td>
</tr>
<tr>
<td>Configuration Reset on the phone UI</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Headset Call Management Records (CMR)</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X*</td>
</tr>
</tbody>
</table>

- * This feature is only available on headsets with Headset Firmware 1.5 or later.
- N/A When you upgrade to Unified CM 12.5(1)SU1 from an earlier version, most Cisco IP Phones will upgrade automatically to Phone Firmware 12.5(1)SR3.

### Table 59: Headset Serviceability Features for Cisco Jabber

<table>
<thead>
<tr>
<th>New Serviceability Feature</th>
<th>Unified CM 12.5(1) or earlier + Jabber version 12.5(1) or earlier</th>
<th>Unified CM 12.5(1)SU1 or earlier + Jabber version 12.5(1) or earlier</th>
<th>Unified CM 12.5(1) or earlier + Jabber version 12.6(1)</th>
<th>Unified CM 12.5(1)SU1 or earlier + Jabber version 12.6(1)</th>
<th>Unified CM 12.5(1) or earlier + Jabber version 12.6(1)MR</th>
<th>Unified CM 12.5(1)SU1 or earlier + Jabber version 12.6(1)MR</th>
</tr>
</thead>
<tbody>
<tr>
<td>COP file installation</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Headset firmware management through Unified CM</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>Remote headset configuration through Unified CM</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X</td>
<td>—</td>
<td>X</td>
</tr>
<tr>
<td>Headset inventory on Unified CM</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X*</td>
<td>—</td>
<td>X*</td>
</tr>
<tr>
<td>Local configuration reset</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X</td>
<td>X</td>
</tr>
</tbody>
</table>
**Workflow: Configure Headset Serviceability**

Use the following workflow to guide you through the setup of your Cisco Headset Serviceability feature.

After you complete this workflow, you can configure headset settings, maintain headset latest firmware loads, and maintain inventory.

**Procedure**

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Activate Cisco Headset Service, on page 661</td>
<td>Turn on <strong>Cisco Headset Service</strong> in Cisco Unified Serviceability.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Prepare Your Headset COP Files, on page 662</td>
<td>Make sure you install and upgrade the latest headset firmware using a COP file.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a User Profile for Your User, on page 663</td>
<td>Create a user profile to be associated with your end user and their associated devices.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Apply User Profiles to End Users, on page 664</td>
<td>Apply the user profiles created for each of your end users through their Cisco IP Phones.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure a Headset Configuration Template, on page 669</td>
<td>Customize the default settings for Cisco headsets. This step is especially important when you deploy the various Cisco Headset series.</td>
</tr>
<tr>
<td>Step 6</td>
<td>View Headset Inventory, on page 671</td>
<td>Check that you can see your deployed headset inventory through Cisco Unified CM Administration interface.</td>
</tr>
</tbody>
</table>

**Activate Cisco Headset Service**

Before you can begin administering Cisco Headsets through the Cisco Unified CM Administration interface, turn on **Cisco Headset Service** in Cisco Unified Communications Manager Serviceability.

---

**New Serviceability Feature**

<table>
<thead>
<tr>
<th>New Serviceability Feature</th>
<th>Unified CM 12.5(1) or earlier + Jabber version 12.5(1) or earlier</th>
<th>Unified CM 12.5(1)SU1 + Jabber version 12.5(1) or earlier</th>
<th>Unified CM 12.5(1) or earlier + Jabber version 12.6(1)</th>
<th>Unified CM 12.5(1)SU1 + Jabber version 12.6(1)MR</th>
<th>Unified CM 12.5(1)SU1 or earlier + Jabber version 12.6(1)MR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local UI configuration</td>
<td>—</td>
<td>—</td>
<td>X</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>Local headset version display</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>X</td>
</tr>
</tbody>
</table>

* This feature can only detect headsets with Headset Firmware 1.5 or later.
Ensure that you run the Cisco Headset Service on all nodes where the Cisco CallManager service is activated.

Procedure

Step 1
From Cisco Unified CM Administration, navigate to Cisco Unified Serviceability and click Go.

Step 2
Select Tools > Service Activation.

Step 3
Check the Cisco Headset Service check box from the CM Services section and select Save.

What to do next
Prepare your Headset COP Files.

Prepare Your Headset COP Files

You can install and upgrade the latest headset firmware using a COP file. A headset COP file contains all the firmware versions of different headset models along with their configuration data.

Ensure that the Cisco Headset service is up and running before the COP file is installed.

1. Install or upgrade the COP file to the Unified Communications Manager system before you can start using your Cisco headsets.

2. Restart the TFTP services once you install the COP file on the system.

When you connect your headset to the endpoints, the headset template configuration changes are applied. If you make any updates to the headset template configurations on Unified Communications Manager, the endpoints apply these configuration updates on the connected headsets.

All configuration updates depend on the version of the headset template in the COP file. If the headset template version is higher in the latest COP file, the configuration file on Unified Communications Manager is updated. If the configuration file in the COP file is upgraded, the headset template version in Unified Communications Manager is updated irrespective of the version of the template and vice versa. The following table shows the various template version update scenarios after a COP file upgrade:

|-----------------------------------------------|----------------------|----------------------------------------------------------|
Configure a User Profile for Your User

Assign a headset template to your users through the User Profile. Configure multiple user profiles for different groups of users as per your deployment needs.

Procedure

Step 1 From Cisco Unified CM Administration, choose User Management > User Settings > 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 If you want Cisco Jabber users associated with this user profile to be able to use the Mobile and Remote Access (MRA) feature, check the Enable Mobile and Remote Access check box.
   Note By default, this check box is selected. When you uncheck this check box, the Jabber Policies section is disabled and No Service client policy option is selected by default.
   Note This setting is mandatory only for Cisco Jabber users. Non-Jabber users do not need this setting to be able to use MRA. The MRA feature is applicable only for Jabber MRA users and not to any other endpoints or clients.
Step 8 Assign the Jabber policies for this user profile. From the Jabber Desktop Client Policy, and Jabber Mobile Client Policy drop-down list, choose one of the following options:
   • No Service—This policy disables access to all Cisco Jabber services.
   • IM & Presence only—This policy enables only instant messaging and presence capabilities.
   • IM & Presence, Voice and Video calls—This policy enables instant messaging, presence, voicemail, and conferencing capabilities for all users with audio or video devices. This is the default option.
   Note Jabber desktop client includes Cisco Jabber for Windows users and Cisco Jabber for Mac users. Jabber mobile client includes Cisco Jabber for iPad and iPhone users and Cisco Jabber for Android users.
Step 9 If you want the users in this user profile to set the maximum login time for Extension Mobility or Extension Mobility Cross Cluster through the Cisco Unified Communications Self Care Portal, check the Allow End User to set their Extension Mobility maximum login time check box.
   Note By default Allow End User to set their Extension Mobility maximum login time check box is unchecked.
Step 10  
Click Save.

Apply User Profiles to End Users

Configure a user profile and associate it with a call device if you have not already done so. This user profile must be associated with the end user, and the MAC of the device must be added under the controlled devices to apply the headset template configuration changes.

Procedure

Step 1  
To add a new end user to the Unified Communications Manager database manually, perform the following:

a) In Cisco Unified CM Administration, choose User Management > End User.

b) Click Add New.

c) Enter the UserID and Lastname.

d) Choose the User Rank from the drop-down list.

e) Complete the fields in the End User Configuration window. For field descriptions, see the online help.

f) Click Save.

Step 2  
To associate the end user with the device, perform the following:

a) In Cisco Unified CM Administration, choose Device > Phone.

b) Select the Cisco IP Phone or device.

c) Under Device Information, select User as the Owner and select the Owner User ID.

d) Click Save and Apply Config for the configuration changes to take effect.

Headset Template Management

You can assign headset templates to user profiles in Cisco Unified Communications Manager to configure default headset settings for your users. The headset template provides the option to associate User Profiles. Unified Communications Manager supports two types of headset templates:

Standard Default Headset Configuration Template

This is the system default template that contains the factory default settings for all headset model series. This template contains the headset settings supported by the latest headset firmware installed on your system for all your headset model series. You cannot edit the default settings though you can change the profile configuration setting.

Note

The Standard Default Headset Configuration template is created only when the Cisco Headset Service is activated in the Cisco Unified Serviceability user interface.

By default, all User Profiles are associated to the standard headset template unless the administrator associates these user profiles to any of the custom defined headset templates. You can make copies of the standard default
headset template to create custom template with customized values of the parameters including the headset firmware version.

**System Generated Custom Headset Template**

Before release 12.5(1)SU1, administrators used to control headsets using the hand coded configuration file (defaultheadsetconfig.json file) that was manually uploaded to the Cisco TFTP server. After upgrade to Unified CM Release 12.5(1)SU1 or later, the defaultheadsetconfig.json file (which includes the headset settings and firmware version) is converted to a custom headset template. This template is displayed as **System Generated Custom Headset Template** in the Cisco Unified CM Administration interface. All the existing users and devices are still associated to this custom template after the upgrade.

**Custom Headset Configuration Template**

You can create a customized headset template as per your deployment needs. You can assign different headset parameters to different models in the same template. You can also assign different firmware loads to different headset models. The custom headset settings can be assigned to specific sets of users by associating the User Profile(s) to the Custom Headset Template.

**Table 60: Headset Template Configuration Settings**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Headset Template Configuration</strong></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>Enter a unique name to identify the headset template.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description that identifies use of the template.</td>
</tr>
<tr>
<td><strong>Model and Firmware Settings</strong></td>
<td></td>
</tr>
<tr>
<td>Choose Model Series</td>
<td>Choose any supported headset model that offers reliable, high-quality sound for your device.</td>
</tr>
</tbody>
</table>
| Add                 | For a standard template, you can view the default pre-defined firmware versions and settings of the headset models. You cannot edit the default values.  
                        For customized templates, click **Add** to add a new headset model and corresponding settings. You cannot add another existing headset model in the same template. You can add different headset models in a customized template; however, you can only use one firmware per headset model. For more information on headset parameters, see the "Headset Configuration Parameters" table below.  
                        For Standard Default Headset Template Configuration, you can only edit settings by installing a headset COP file. |
### Field | Description
--- | ---
Firmware | Select the required firmware version.  
- **Remain on current version**—Choose this option if you want the headset to remain on the existing firmware version (that is, the headset firmware version is not upgraded to the latest firmware version on the system).  
- **Latest**—Choose this option if you want to upgrade the headset firmware version to the latest firmware version on the system.

Delete | For customized templates, click **Delete** to remove the headset model from the headset template.

### Profile Configuration

#### Available User Profiles

Lists the configured User Profiles that are available to use with this headset template.  
To associate a User Profile to this template, select the profile and click the down arrow to move the template to Assigned User Profiles.  

*Note*  
By default, all User Profiles get assigned to the Standard Default Headset Configuration Template. To associate a User Profile to a different template, create the new template and assign the User Profile to the new template.

#### Assigned User Profiles

Lists the User Profiles that will use this headset configuration template. For users assigned to this profile, the settings in this headset configuration template are applied to their Cisco headsets during registration.  
Click the arrows to add new User Profiles from the Available User Profiles list.

---

The following table describes the parameters in each headset template.
Table 61: Headset Configuration Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Range</th>
<th>Default</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Speaker Volume</td>
<td>0 – 15</td>
<td>7</td>
<td>Controls the level of sound in the headset. 0 is very low while 15 is loud. Configure this setting based on the ambient noise in the office environment. This parameter applies to all headsets in the Cisco Headset 500 Series.</td>
</tr>
<tr>
<td>Microphone Gain</td>
<td>−6 dB – +6 dB</td>
<td>0</td>
<td>Gain controls how loud the user sounds to other people on the call. −6 dB means users sound quiet while +6 dB means users sound much louder. Configure this setting based on the ambient noise in the office environment. This parameter applies to all headsets in the Cisco Headset 500 Series.</td>
</tr>
<tr>
<td>Sidetone</td>
<td>Off – High</td>
<td>Low</td>
<td>Controls how much of a user’s own voice they can hear through their headset. Off turns off the sidetone while High means that users receive much more feedback from their headset microphones. This parameter applies to all headsets in the Cisco Headset 500 Series.</td>
</tr>
<tr>
<td>Equalizer</td>
<td>Warmest – Brightest</td>
<td>Default</td>
<td>Controls the Equalizer settings. Warmer settings mean users hear more bass in their headsets, while a brighter setting means users hear more treble. This parameter applies to all headsets in the Cisco Headset 500 Series.</td>
</tr>
<tr>
<td>Audio Bandwidth</td>
<td>Wide Band, Narrow Band</td>
<td>Wide Band</td>
<td>Controls the Digital Enhanced cordless Telecommunications (DECT) codec in the Cisco Headset 560 Series. In a dense DECT environment, set the field to Narrow Band to limit the Cisco Headset 560 Series to the G.726 codec.</td>
</tr>
<tr>
<td>Parameter</td>
<td>Range</td>
<td>Default</td>
<td>Notes</td>
</tr>
<tr>
<td>---------------------</td>
<td>----------------</td>
<td>---------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Bluetooth</td>
<td>On, Off</td>
<td>On</td>
<td>Controls the use of Bluetooth on the Cisco Headset 560 Series with Multibase. When this parameter is set to <strong>Off</strong>, the base deletes all devices paired with it. <strong>Note</strong> You can disable Bluetooth if your users are operating in an environment in which information security is at a premium.</td>
</tr>
<tr>
<td>Conference</td>
<td>On, Off</td>
<td>Off</td>
<td>Controls the use of the conferencing feature on the Cisco Headset 560 Series. Conferencing allows up to three guest headsets to pair with the same base at once. <strong>See Cisco Headset 500 Series User Guide</strong> for more information on conferencing.</td>
</tr>
<tr>
<td>Firmware Source</td>
<td>Allow from Unified Communications Manager or Cisco Cloud (firmware will upgrade only), Restrict to Unified Communications Manager only (firmware may upgrade or downgrade)</td>
<td>Allow from Unified Communications Manager or Cisco Cloud (firmware will upgrade only)</td>
<td>Controls the headset's firmware upgrade source. By default, users can upgrade their headset through a Cisco IP Phone connected to Unified Communications Manager or through Cisco Jabber for Windows or Mac. You can restrict your headsets to only accept firmware changes through a Unified Communications Manager source.</td>
</tr>
<tr>
<td>DECT Radio Range</td>
<td>Autorange, Medium Range, Short Range</td>
<td>Autorange</td>
<td>Controls the maximum distance between the Cisco Headset 560 Series and its base. By default, the bases have a DECT range of over 330 feet (100 meters) in ideal conditions. If you configure the DECT radio range to <strong>Medium Range</strong> or <strong>Short Range</strong>, the headset base consumes less power but users can't move as far from the base while on a call. Configure DECT radio range to <strong>Short Range</strong> for high density headset deployment. <strong>For more detailed information on DECT deployment, refer to the white paper on Cisco Headset deployment, How to Deploy DECT at Work for the Cisco Headset 560 Series.</strong></td>
</tr>
</tbody>
</table>
Configure a Headset Configuration Template

Use this procedure to create a template for your Cisco headset settings. You can create a customized template or use the system-defined Standard Default Headset Template.

Note

The Standard Default Headset Configuration Template is a system-defined template. You can assign new User Profiles to the Standard Default Headset Template but you can't edit the template. By default, all user profiles are assigned to this template. To disassociate a user profile from this template, you must assign the profile to a new template.

Procedure

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

**Step 2**
Do either of the following:
- To edit an existing template, select the template.
- To create a new template, select any existing template and click Copy. The existing settings are applied to your new template.

**Step 3**
Add a Name and Description for the template.

**Step 4**
Under Model and Firmware Settings, assign any customized headset settings that you want to apply to this template. To add a new setting, click the Add button and configure the settings.

**Step 5**
Assign users to this template by using the up and down arrows to move their profiles to the Assigned User Profiles list.

**Step 6**
Click Save.

**Step 7**
Use the Set to Default button to return to the default template settings.

**Step 8**
Click Apply Config.

For a Standard Default Headset Configuration Template, the Apply Config button takes effect for the following:
- Devices owned by users you added to the Assigned User Profile list
- Anonymous devices

For a Customized Headset Configuration Template, the Apply Config button takes effect only for devices owned by users you added to the Assigned User Profile list.

Firmware Management

Most phones and devices connected to the Unified Communications Manager support the Cisco Headset 500 Series. Install the latest phone firmware release and device package before connecting your headset to a phone. When the headset first connects, it downloads the required firmware and begins the upgrade process.

For a given headset model, the following two firmware options are supported:
• **Remain on current version**—Choose this option if you want the headset to remain on the existing firmware version (that is, the headset firmware version is not upgraded to the latest system firmware version).

• **Latest**—Choose this option if you want to make the headset firmware version to match the latest firmware version on the system.

For example, during upgrade:

- If you choose Latest (1-5-1-10) as your chosen firmware option and the existing Headset firmware is (1-5-1-9), an upgrade will occur.
- If you choose Latest (1-5-1PA-10) as your chosen firmware option and the existing Headset firmware is (1-5-1-11), an upgrade (in this case downgrade) will occur.
- If you choose Latest (1-5-1-10) as your chosen firmware option and the existing Headset firmware is (1-5-1-11), an upgrade (in this case a downgrade) will not occur.

---

**Note**

If you choose **Latest 1-5-1-10** as your chosen firmware option and the existing Headset firmware is **1-5-1-11**, that turns out to be an older version than the one on the headset, upgrade (in this case a downgrade) will occur. This is a known limitation.

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**Firmware Considerations**

- Users assigned to the standard headset template will always receive the latest headset firmware and settings.
- Settings shown in the Headset Template Configuration (both Standard and Custom) are always set to the **Latest firmware** for all headset model series.

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**Headset Inventory Management**

Cisco IP Phones send headset inventory data to Unified Communications Manager whenever the headset is in a connected or disconnected state. Unified Communications Manager stores the inventory data so you can generate an Inventory Summary Report or Custom Inventory Report for all headsets deployed in this server. Report information includes: headset serial and model number, docking station details, firmware, configuration templates used, vendor details, and headset connection status to devices.

---

**Headset Inventory Settings**

You can view a full list of all headsets deployed on your server. You can use this information to generate reports for all deployed headsets. If you click the Serial Number of the device, you can view details of individual headsets in a pop-up window.
Table 62: Headset Inventory Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Serial Number</td>
<td>Serial Number of the headset. This number is unique for every individual headset.</td>
</tr>
<tr>
<td>Note</td>
<td>For non-Cisco headsets, the Device Name is used as the Serial Number. Using the same non-Cisco headset with multiple phones creates duplicate headset records.</td>
</tr>
<tr>
<td>Model</td>
<td>Model number of the headset.</td>
</tr>
<tr>
<td>Vendor</td>
<td>Displays vendor details.</td>
</tr>
<tr>
<td>Type</td>
<td>Indicates the type of flexible options for connecting the headset to devices: USB, RJ-3.5 mm, DECT, and Bluetooth connections.</td>
</tr>
<tr>
<td>Firmware</td>
<td>Displays the most current firmware load of the headset.</td>
</tr>
<tr>
<td>User</td>
<td>Displays information of the end user using this particular headset.</td>
</tr>
<tr>
<td>Template</td>
<td>Display the name of the headset configuration template.</td>
</tr>
<tr>
<td>Status (since)</td>
<td>Displays the status of the headset activities. It can be: Connected or Disconnected.</td>
</tr>
<tr>
<td>Dock Model</td>
<td>Displays the type of docking model station.</td>
</tr>
<tr>
<td>Device Name</td>
<td>Name of the device to which the headset is connected to.</td>
</tr>
<tr>
<td>Device Model</td>
<td>Displays the Cisco IP Phone or Cisco Jabber model number. For example, CP-8865 is a Cisco IP Phone model. CSF is a device type for either Cisco Jabber for Mac or Cisco Jabber for Windows.</td>
</tr>
<tr>
<td>Software Version</td>
<td>Displays the latest version of the software used. It can be a phone firmware or a Jabber software version.</td>
</tr>
<tr>
<td>Headset Age (days)</td>
<td>Displays the age of the headset after getting connected to the Unified Communications Manager server. If the record is deleted, the headset age is reset.</td>
</tr>
</tbody>
</table>

View Headset Inventory

You can view a full list of all headsets deployed on your server. You can use this information to generate reports for all deployed headsets.
Procedure

**Step 1**
From Cisco Unified CM Administration, select **Device > Headset > Headset Inventory**.

**Step 2**
Do either of the following:

- Select **Find** to see a full list of headsets deployed on your server.
- Enter one or more search criteria into the search box and select **Find**.

---

**Headset Inventory Summary**

You can view an aggregate summary of your deployed headsets in the Headset Inventory Summary window.

**Headset Inventory by Model**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Headset Model</td>
<td>The headset model number.</td>
</tr>
<tr>
<td>Quantity</td>
<td>Lists the number of headsets for each model type in your deployment.</td>
</tr>
</tbody>
</table>

**Note** Click the link in the Quantity column to navigate to the detailed Headset Inventory page with filter set to the model type selected.

**Headset Inventory by Status**

**Note**
Click the hyperlinks in the Headset Model, Active, Inactive, and Unassigned columns in the following table to navigate to the detailed Headset Inventory page with prefiltered status.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Headset Model</td>
<td>The headset model number.</td>
</tr>
<tr>
<td>Active</td>
<td>The status is active when the headset is connected or disconnected within the last 30 days.</td>
</tr>
<tr>
<td>Inactive</td>
<td>The status is inactive when the headset is not connected in the last 30 days.</td>
</tr>
<tr>
<td>Unassigned</td>
<td>The status is unassigned when the user ID does not exist in the system or the inventory record doesn't have a user ID mapping.</td>
</tr>
</tbody>
</table>
Get an Aggregate Summary of Your Deployed Headsets

You can view an aggregate summary of your deployed headsets in the Headset Inventory Summary window.

**Procedure**

In Cisco Unified CM Administration, select **Device > Headset > Headset Inventory Summary**.

You can view a breakdown of headset inventory by model or by headset status.

Headset Troubleshooting and Diagnostics

You can configure Unified Communications Manager or Cisco Unified Real-Time Monitoring Tool (RTMT) to collect Problem Report Tool (PRT) logs for headsets connected to Cisco IP Phones. The PRT includes data on call quality, codecs used, audio settings, wireless settings, and alert logs.

Unified Communications Manager stores the call diagnostics details for Headsets. Cisco IP Phones send headset diagnostics data in Headset-Stat header either in a BYE message or a 200 OK response to BYE message to update the CMRs in Unified Communications Manager.

Cisco IP Phones share the headset diagnostics data with Unified Communications Manager and this information is stored in the CMR records as two new fields:

- **headsetSN**—Serial number of the headset.
- **headsetMetrics**—Headset metrics such as RSSI frame errors, connection drop reason, beacon moves, audio settings, and DECT bandwidth.

For more information on the new CMR records, see the Call Detail Records Administration Guide for Cisco Unified Communications Manager, Release 12.5(1)SU1.

Generate PRT for Endpoints on Unified CM

Use this procedure to trigger the Problem Reporting Tool (PRT) on the endpoints.

**Procedure**

1. From Cisco Unified CM Administration, choose **Device > Phone**.
2. Click **Find** and select one or more phones that the headset connects to.
3. Click **Generate PRT for Selected** to collect PRT logs for the headsets used by the selected phones.
4. Click **Save**.

Unified Communications Manager sends SIP Notify messages to remotely trigger the log collection on the phone and upload it to the log server configured in the “Customer support upload URL” parameter.
Generate PRT for Endpoints on RTMT

Devices or endpoints generate alarms for each critical event for diagnostics and troubleshooting. These alarms are generated using the Problem Reporting Tool (PRT) available in the Trace Collection menu or Device Monitoring menu of the Cisco Unified Real-Time Monitoring Tool (RTMT) user interface.

Procedure

Step 1  
Open the Trace and Log Central options.

Step 2  
In the Trace & Log Central tree hierarchy, choose Generate PRT.  
The Generate PRT wizard appears.

Step 3  
Enter the Device name as configured in the Find and List Phones page in the Cisco Unified CM Administration user interface.

Step 4  
Click Generate PRT.  
The generated report is uploaded at the Customer support upload URL. The download option is available only if the Customer support upload URL parameter is configured at the Enterprise, Profile, or Device level in the Cisco Unified CM Administration user interface.

Note  
Check the Customer support upload URL parameter in the Enterprise, Profile, or Device level configuration page settings. Else, PRT generation fails.
Video Endpoints Management Overview

This feature simplifies the administrator's job of provisioning and managing Cisco TelePresence video endpoints. An administrator can provision settings for Cisco TelePresence endpoints in Unified Communications Manager and then push those Product-Specific Configuration settings to endpoints.

Prior to Release 12.5(1)SU1, only a limited set of Product-Specific Configurations were pushed from Unified Communications Manager to the endpoint resulting in a partial configuration of the endpoint. Administrator had to rely on Cisco TelePresence Management Suite or TelePresence Endpoint's web interface to configure all the settings. The Phone Configuration window in Unified Communications Manager contains a complete Product-Specific Configuration layout for Cisco TelePresence endpoints that matches what users see on their endpoint. This update lets administrators apply settings on behalf of users and then push those settings to users.

The Bulk Administration Tool (BAT) Phone Template Configuration page also displays the new model-specific configurations in a tabbed layout, supporting the complete list of endpoint parameters. You can import the entire set of parameters or modify a specific parameter in the endpoint in bulk.

Video endpoints management features provide the following benefits:

- TelePresence endpoints can be fully provisioned from Unified Communications Manager—Endpoints parameters listed in the Unified Communications Manager user interface are in the same order as listed in the Advanced Configuration settings of your Cisco TelePresence model. For more information on the various advanced parameters, see the respective model in the Collaboration Endpoints Administrator Guides.

- New Product-Specific Configuration Layout—New layout details the model-specific configurations in a tabbed layout. This is an upgrade from the earlier flat format that provided access only to a limited
set of parameters. The new layout ensures that you have a complete list of Cisco TelePresence settings on the Cisco Unified CM Administration interface.

- Automatic migration of the configuration data from the video endpoints—This simplifies the deployment of endpoints by automatically synching data from endpoints to Unified Communications Manager and vice versa. Endpoint configurations can be fully restored in case of reset to factory settings or Product Returns & Replacements (RMA) swaps.

---

**Video Endpoints Management Feature Compatibility**

Following table details the video endpoints management feature compatibility with Unified Communications Manager and Collaboration Endpoint (CE) versions:

<table>
<thead>
<tr>
<th>Unified Communications Manager Version</th>
<th>CE Endpoint Version</th>
<th>Expected Behavior</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.5(1) SU1</td>
<td>9.8 and above</td>
<td>Devices added prior to 12.5(1) SU1:</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Advanced Configuration UI (Tabbed Layout) for successfully backed up devices</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Limited Configuration UI (Flat Layout) for devices yet to be backed up</td>
</tr>
<tr>
<td></td>
<td></td>
<td>New device added through UI/BAT/AXL:</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Advanced Configuration UI</td>
</tr>
<tr>
<td>12.5(1) SU1</td>
<td>9.7 and below</td>
<td>Devices added prior to 12.5(1) SU1:</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Limited Configuration UI</td>
</tr>
<tr>
<td></td>
<td></td>
<td>New device added through UI/BAT/AXL:</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- Advanced Configuration UI with only a limited set of parameters taking into effect</td>
</tr>
<tr>
<td>12.5(1) and below</td>
<td>9.8 and above</td>
<td>Limited Configuration UI</td>
</tr>
</tbody>
</table>

Note

Any endpoint that supports Collaboration Endpoint (CE) Software 9.8 or higher can use this new provisioning layout for the Product-Specific Configuration fields on the Phone Configuration page. If you are using a CE software version prior to 9.8, you will be able to view all the new set of advanced parameters; but, the new set of parameters functions only if you upgrade your CE Software version to 9.8 or higher. The subset of parameters supported is marked with a “#” to the right of each parameter value in the user interface. You must load a device pack onto Unified Communications Manager if a device type is capable of supporting the new provisioning framework, but does not show the additional parameters.
Migration Considerations for Video Endpoints Provisioning

Auto Backup After Unified Communications Manager Upgrade

When upgrading to Unified Communications Manager 12.5(1)SU1, the existing configuration data for the supported endpoint types is automatically migrated from endpoints to Unified Communications Manager.

1. Upgrade Unified Communications Manager to version 12.5(1)SU1 or later.
2. Endpoints register to Unified Communications Manager.
3. Unified Communications Manager then sends a SIP Notify message to endpoints requesting for the full set of Product-Specific Configuration parameters.
4. Endpoints that are upgraded to CE 9.8 and above send full set of configuration data to Unified Communications Manager (in xConfiguration format) using a SIP REFER message.
5. Unified Communications Manager processes this configuration data and populates the complete list of Cisco TelePresence settings (Advanced Configuration UI) on the Cisco Unified CM Administration interface.

Note

Unified Communications Manager server displays the complete endpoint configuration settings in the new layout only if Unified CM is able to successfully back up data from the endpoint.

Configuration Control Modes

Based on the deployment needs, administrators can configure various configuration control modes in the Cisco Unified CM Administration interface. You can decide whether you want to control the configuration settings centrally from the endpoints or Unified Communications Manager or both of them together.

Navigate to the Product-Specific Configuration Layout section on the Phone Configuration page and choose the Configuration Control Mode under “General Settings” in the Miscellaneous tab to control the various modes. Following are the various Configuration Control Modes:

- **Unified CM and Endpoint (Default)**—Use this mode if you want Unified Communications Manager and endpoint to operate as the multi-master source for provisioning endpoint data. If Unified CM and Endpoint is the configured mode, any update made via an endpoint locally is synched with the Unified CM server.
- **Unified CM**—Use this mode if you want Unified Communications Manager to operate as the centralized master source for provisioning endpoint data and does not want to accept any configurations done from the endpoints locally.
- **Endpoint**—Use this mode if you want endpoints to operate as the centralized master source of configuration data. In this mode, endpoint ignores any configuration data from Unified Communications Manager and doesn’t sync back the changes done locally. This mode is typically used when an Audiovisual (AV) integrator is installing the endpoints and wants to control configuration from the endpoint.
In the **Endpoint** mode, CE devices continue to accept that limited set of parameters supported prior to release 12.5(1)SU1. Unified Communications Manager indicates these parameters with a "#" symbol. CE devices will ignore the extended set of parameters supported from the 12.5(1)SU1 release onwards.

---

**On-demand Configuration Pull Functionality**

Administrators can use the **Get Config from Phone** option to pull configuration changes from the CE 9.8 endpoint devices on-demand at that given point.

Navigate to the Product-Specific Configuration Layout section on the Phone Configuration page and click the **Get Config from Phone** button on the top corner of the page to pull any data configuration from the CE 9.8 endpoints on-demand. This option is enabled only if the endpoint is in the registered state.

---

**Video Endpoints Migration Report**

**Video Endpoint with Extended Configuration Backup** is the new filter is introduced on the Find and List Phones window for release 12.5(1)SU1. Administrators can search for details on how many CE endpoints got migrated automatically and how many CE endpoints did not. Based on this information, they can take corrective measures.

---

**Provisioning and Migration Scenarios**

The following table describe various provisioning and migration scenarios. All of these scenarios assume that your TelePresence video endpoints are upgraded to a CE release that supports Product-Specific Configuration provisioning from Unified CM. In Unified CM, these settings appear in the **Product-Specific Configuration** section, but on the endpoint, they appear under **Advanced Configuration**.

### Table 63: Provisioning and Migration Scenarios for Video Endpoints

<table>
<thead>
<tr>
<th>Task</th>
<th>Existing Configuration Summary</th>
<th>What to do</th>
</tr>
</thead>
<tbody>
<tr>
<td>Provisioning New Video Endpoints</td>
<td>• Brand new device</td>
<td>With Unified CM at a minimum release 12.5(1)SU1 and the CE endpoint at 9.8, you can provision new endpoints and manage the product-specific configurations from Unified CM.</td>
</tr>
<tr>
<td></td>
<td>• Device is not provisioned on Unified CM</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• No existing settings on the device or on Unified CM</td>
<td></td>
</tr>
</tbody>
</table>
### Add Migrating Video Endpoint to Unified CM

If you are migrating existing Cisco TelePresence Video Endpoints from a Cisco TelePresence Video Communications Server to Unified Communications Manager, use this procedure to add the CE endpoint into Unified CM via the **Phone Configuration** window so that the existing **Advanced Configurations** from the endpoint can be managed from the **Phone Configuration** window in Unified CM.

<table>
<thead>
<tr>
<th>Task</th>
<th>Existing Configuration Summary</th>
<th>What to do</th>
</tr>
</thead>
</table>
| Migrating Existing Video Endpoints from VCS | • Existing device  
• Device is not provisioned on Unified CM  
• Device is configured, but Unified CM does not have any of the configurations | If you are migrating existing video endpoints from a Cisco TelePresence Video Communications Server to Cisco Unified Communications Manager:  
**Adding Phones via Phone Configuration window in Unified CM:**  
• Add the phone to Unified CM, but **DO NOT CLICK Save.**  
• Register the phone. After registration, the existing **Advanced Configuration** settings from the phone are uploaded to Unified CM and display in **Product-Specific Configurations** in the **Phone Configuration** window.  
• In the **Phone Configuration** window, configure the new settings and click **Save.** The provisioned settings download to the phone.  
For a detailed procedure, see **Add Migrating Video Endpoint to Unified CM**, on page 679  
**Adding Phones via Bulk Administration**  
Make sure that the csv file or BAT Template that you use for provisioning does not include the Product-Specific Configuration fields.  
**Adding Phones via AXL**  
Make sure that the AXL request does not include any Product-Specific Configuration fields. |
| Upgrading from an Earlier Release of Unified CM with Registered Video Endpoints | • Existing device  
• Device is provisioned on a pre-12.5 release of Unified CM  
• Unified CM has a limited set of Product-Specific Configuration settings for the device | So long as the CE endpoint is at a supported version, when you upgrade Unified CM, the **Advanced Configuration** settings from the endpoint get pulled into Unified CM automatically following device registration and display under the **Product-Specific Configuration** section of the **Phone Configuration** window.  
After registration, you can set the **Configuration Control Mode** in addition to whatever settings you want. |
Make sure to follow this procedure closely. The settings from the endpoint do not automatically upload to Unified CM until after device registration.

This procedure uses the Add New from Template setting in the Unified CM Phone Configuration window. You can also use tools like Bulk Administration or AXL to add the endpoint.

Procedure

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

Step 2
Click Add New from Template and enter the following phone details:

- Select the model from the Phone Type drop-down.
- Enter the MAC Address of the endpoint.
- From the Device Template, select a Universal Device Template.
- Select the Directory Number that you want to add to the phone. If none exists, click New and configure a directory number.
- From the User drop-down, select the user whom will own the device.

Step 3
Click Add. The Phone Configuration displays with the universal device template settings filling out the phone configuration. The Product-Specific Configuration section also appears, but with default settings, rather than the existing settings from the phone.

Note: You can also add the device using the Phone Configuration window's Add New button, but this method requires that you enter settings manually.

Step 4
DO NOT CLICK Save. If you save settings, Unified CM does not load existing settings from the phone. If you saved by mistake, go straight to the troubleshooting Note at the bottom of this procedure for recovery steps.

Step 5
Register the phone. During registration, the existing Advanced Configuration settings from the phone get pulled into Unified CM and display in the Phone Configuration window’s Product-Specific Configuration section.

Step 6
In the Phone Configuration window, configure how you want endpoint settings to be managed by configuring the Configuration Control Mode field:

- Unified CM and Endpoint (Default)—Use this mode if you want Unified Communications Manager and endpoint to operate as the multi-master source for provisioning endpoint data. If Unified CM and Endpoint is the configured mode, any update made via an endpoint locally is synched with the Unified CM server.
- Unified CM—Use this mode if you want Unified Communications Manager to operate as the centralized master source for provisioning endpoint data and does not want to accept any configurations done from the endpoints locally.
- Endpoint—Use this mode if you want endpoints to operate as the centralized master source of configuration data. In this mode, endpoint ignores any configuration data from Unified Communications Manager and doesn’t sync back the changes done locally. This mode is typically used when an Audiovisual (AV) integrator is installing the endpoints and wants to control configuration from the endpoint.
Step 7
Configure any phone settings that you want. For help with the fields, refer to the online help.

Step 8
Click Save.
The provisioned settings in Unified Communications Manager download to the endpoint.

Note
If you clicked Save mistakenly in the Phone Configuration window prior to device registration, the existing Advanced Configuration settings from the endpoint will not load to Unified CM when the device registers. To recover, do the following prior to device registration:

- In Unified CM, set the Configuration Control Mode to Endpoint and click Save.
- Let the phone register to Unified CM.
- After registration, return to the device configuration in the Phone Configuration window and click the Get Config from Device button. The setting results in the existing Advanced Configurations on the phone getting pulled into Unified CM. Note that this button does not appear until after device registration.
- Return to Step 6 of the procedure in order to complete the configuration.
PART XIII

Advanced Call Processing

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- Configure External Call Control, on page 695
- Configure Call Queuing, on page 705
- Configure Call Throttling, on page 717
- Configure Logical Partitioning, on page 721
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CHAPTER 55

Configure Call Control Discovery

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- Call Control Discovery Prerequisites, on page 685
- Call Control Discovery Configuration Task Flow, on page 685
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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

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</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</td>
<td>Configure a Cisco IOS router as the SAF forwarder.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td>determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform.</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>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure SAF Security Profile, on page 687</td>
<td>Configure SAF Security Profile, on page 687</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure SAF Forwarders, on page 688</td>
<td>Configure SAF Forwarders, on page 688</td>
<td></td>
</tr>
<tr>
<td>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.</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>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure SIP or H.323 Intercluster Trunks, on page 688</td>
<td>Configure SIP or H.323 Intercluster Trunks, on page 688</td>
<td></td>
</tr>
<tr>
<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>
<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>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure Hosted DN Groups, on page 689</td>
<td>Configure Hosted DN Groups, on page 689</td>
<td></td>
</tr>
<tr>
<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 with hosted DN groups, which allow you to easily associate multiple patterns to a CCD advertising service.</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 with hosted DN groups, which allow you to easily associate multiple patterns to a CCD advertising service.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> Configure Hosted DN Patterns, on page 689</td>
<td>Configure Hosted DN Patterns, on page 689</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><strong>Step 7</strong></td>
<td>Configure the Advertising Service, on page 690</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Configure the Partition for Call Control Discovery, on page 690</td>
<td>Configure a call control discovery partition to ensure that the learned patterns are inserted into digit analysis under this partition.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>Configure the Requesting Service, on page 690</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td>Block Learned Patterns, on page 691</td>
<td>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.</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/%20go/cfn.)

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Advanced Features > SAF > SAF Security Profile.

**Step 2**
Configure the fields on the SAF Security Profile Configuration window.

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

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

**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.

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

**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.
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**

**Table 64: 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 SAFCCDRequestingService 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>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</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>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.</td>
</tr>
</tbody>
</table>

**Call Control Discovery Restrictions**

All clusters are limited to advertised or learned routes within the same autonomous system (AS).
Configure External Call Control

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](https://developer.cisco.com).

## External Call Control 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 a Calling Search Space for External Call Control, on page 697</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>Step 2</td>
<td>Configure an External Call Control Profile, on page 698</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>Step 3</td>
<td>Assign a Profile to a Translation Pattern, on page 698</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>Step 4</td>
<td>(Optional) Import the Route Server Certificate into the Trusted Store, on page 699</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>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>----------------------------------------------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> (Optional) Export the Self-Signed Certificate to the Route Server, on page 699</td>
<td>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.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> (Optional) Configure the Chaperone Function, on page 700</td>
<td>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.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> (Optional) Configure Customized Announcements, on page 701</td>
<td>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.</td>
<td></td>
</tr>
</tbody>
</table>

**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>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified CM Administration, choose <strong>Call Routing</strong> &gt; <strong>Class of Control</strong> &gt; <strong>Calling Search Space</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click <strong>Add New</strong>.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>In the <strong>Name</strong> field, enter a name.</td>
</tr>
<tr>
<td></td>
<td>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 <strong>Description</strong> field, enter a description.</td>
</tr>
<tr>
<td></td>
<td>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><strong>Step 5</strong></td>
<td>From the <strong>Available Partitions</strong> drop-down list, perform one of the following steps:</td>
</tr>
</tbody>
</table>
• 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.
Step 8  Click Save.

---

### 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.

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose Call Routing > External Call Control Profile.
**Step 2**  Perform one of the following tasks:
  • Click Find and then choose an existing external call control profile from the resulting list to modify the settings for an existing external call control profile, enter search criteria.
  • Click Add New to add a new external call control profile.

**Step 3**  Configure the fields on the External Call Control Profile Configuration window. For more information on the fields and their configuration options, see the system Online Help.
**Step 4**  Click Save.

---

### Assign a Profile to a Translation Pattern

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.

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose Call Routing > Translation Pattern.
**Step 2**  Perform one of the following tasks:
  • Click Find and then choose an existing translated pattern from the resulting list to modify the settings for an existing translated pattern, enter search criteria.
  • Click Add New to add a new translated pattern.

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

Step 5 Click Save.

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.

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.

Export the Self-Signed Certificate to the Route Server

If the route server uses HTTPS, export the 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 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.

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.
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.

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.

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.

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

### Table 65: 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>
<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 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, 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. |
<p>| Call Pickup            | When a phone user tries to pick up a call by using the Call Pickup feature, External Call Control is not invoked; 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.                                                                                         |</p>
<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified Mobility</td>
<td>Unified Communications Manager allows the route decision from the adjunct route server for the following Cisco Unified Mobility features:</td>
</tr>
<tr>
<td></td>
<td>• Mobile Voice Access</td>
</tr>
<tr>
<td></td>
<td>• Enterprise Feature Access</td>
</tr>
<tr>
<td></td>
<td>• Dial-via-Office Reverse Callback</td>
</tr>
<tr>
<td></td>
<td>Unified Communications Manager does not send a routing query for the following Cisco Unified Mobility features:</td>
</tr>
<tr>
<td></td>
<td>• Cell pickup</td>
</tr>
<tr>
<td></td>
<td>• Desk pickup</td>
</tr>
<tr>
<td></td>
<td>• Session handoff</td>
</tr>
<tr>
<td>Conferences</td>
<td>When a phone user creates a conference, External Call Control may be invoked for the primary call and consultative call.</td>
</tr>
<tr>
<td>Directory Numbers</td>
<td>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.</td>
</tr>
<tr>
<td>Do Not Disturb</td>
<td>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, Unified Communications Manager rejects the call.</td>
</tr>
<tr>
<td>Emergency Call Handling</td>
<td>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.</td>
</tr>
<tr>
<td>Transfer</td>
<td>When a phone user transfers a call, External Call Control may be invoked for both the primary call and consultative call. However, Unified Communications Manager cannot enforce any routing rules from the adjunct route server between the party that transfers and the target of the transfer.</td>
</tr>
</tbody>
</table>
## External Call Control Restrictions

### Table 66: External Call Control Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Adding Parties</td>
<td>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 <em>Advanced Ad Hoc Conference Enabled</em> 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 <em>True</em>, other parties can add participants to the conference.</td>
</tr>
<tr>
<td>Call Transfer</td>
<td>The chaperone cannot use the phone to transfer the conference call to another party.</td>
</tr>
<tr>
<td>Conference Log Out</td>
<td>When the chaperone leaves the conference, the entire conference ends.</td>
</tr>
<tr>
<td>Conference Softkey</td>
<td>After the chaperone creates a conference, the <em>Conference</em> softkey, if available, is disabled on the phone.</td>
</tr>
<tr>
<td>Hold</td>
<td>The chaperone cannot use the phone to put the conference call on hold.</td>
</tr>
<tr>
<td>Recording</td>
<td>If the chaperone starts recording before the feature makes a consultative call to the party that will join the conference, Unified Communications Manager suspends recording while the chaperone makes the consultative call; recording resumes after the conference is established.</td>
</tr>
</tbody>
</table>
Configure Call Queuing

- Call Queuing Overview, on page 705
- Call Queuing Prerequisites, on page 706
- Call Queuing Task Flow, on page 707
- Call Queuing Interactions, on page 713
- Call Queuing Restrictions, on page 714
- Performance and Scalability for Hunt Pilots with Call Queuing, on page 714

Call Queuing Overview

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 queuing or nonqueuing 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 queuing-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 queuing-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

### 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 Announcements, on page 707</td>
<td>Configure announcements through uploading .wav files.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Music On Hold, on page 708</td>
<td>Configure Music On Hold (MoH) audio source.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Hunt Pilot Queuing, on page 711</td>
<td>Enables call queuing hold option for the calls in a queue until they are answered.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Automatically Logout Hunt Member on No Answer, on page 713</td>
<td>Allows line members to log off the hunt list automatically.</td>
</tr>
</tbody>
</table>

### 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.
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 709.

**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 67: 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>
<tr>
<td>MOH Audio Source File Status</td>
<td>This pane displays the following information about the source file for the selected MOH audio source:</td>
</tr>
<tr>
<td></td>
<td>• InputFileName</td>
</tr>
<tr>
<td></td>
<td>• ErrorCode</td>
</tr>
<tr>
<td></td>
<td>• ErrorText</td>
</tr>
<tr>
<td></td>
<td>• DurationSeconds</td>
</tr>
<tr>
<td></td>
<td>• DiskSpaceKB</td>
</tr>
<tr>
<td></td>
<td>• LowDateTime</td>
</tr>
<tr>
<td></td>
<td>• HighDateTime</td>
</tr>
<tr>
<td></td>
<td>• OutputFileList</td>
</tr>
<tr>
<td></td>
<td>• MOH Audio Translation completion date</td>
</tr>
</tbody>
</table>

**Note** OutputFileList includes information on ULAW, ALAW, G.729, and Wideband wav files and status options.
Table 68: Announcement Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Initial Announcement</td>
<td>Choose an initial announcement from the drop-down list.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> To select MoH with no initial announcement, choose the <strong>Not Selected</strong> option.</td>
</tr>
<tr>
<td></td>
<td>Click the <strong>View Details</strong> link to view the following Initial Announcement information:</td>
</tr>
<tr>
<td></td>
<td>• Announcement Identifier</td>
</tr>
<tr>
<td></td>
<td>• Description</td>
</tr>
<tr>
<td></td>
<td>• Default Announcement</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> • Played by MOH server only when the Audio Source has “Allow Multi-casting” unchecked and “Initial Announcement Played” set to 'Only for queued calls'.</td>
</tr>
<tr>
<td></td>
<td>• Played by ANN if “Allow Multi-casting” is checked or if “Initial Announcement Played” is set to 'Always.'</td>
</tr>
<tr>
<td>Initial Announcement Played</td>
<td>Choose one of the following to determine when to play the initial announcement:</td>
</tr>
<tr>
<td></td>
<td>• Play announcement before routing to Hunt Member</td>
</tr>
<tr>
<td></td>
<td>• Play announcement if call is queued</td>
</tr>
<tr>
<td>Periodic Announcement</td>
<td>Choose a periodic announcement from the drop-down list.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> To select MoH with no periodic announcement, choose the <strong>Not Selected</strong> option.</td>
</tr>
<tr>
<td></td>
<td>Click the <strong>View Details</strong> link to view the following Periodic Announcement information:</td>
</tr>
<tr>
<td></td>
<td>• Announcement Identifier</td>
</tr>
<tr>
<td></td>
<td>• Description</td>
</tr>
<tr>
<td></td>
<td>• Default Announcement</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> The MOH server always plays the periodic announcement regardless of other settings.</td>
</tr>
<tr>
<td>Periodic Announcement Interval</td>
<td>Enter a value (in seconds) that specifies the periodic announcement interval.</td>
</tr>
<tr>
<td></td>
<td>Valid values are 10 to 300. The default value is 30.</td>
</tr>
</tbody>
</table>
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 69: Music On Hold Audio Sources

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| 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. |

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.

**Note**  
If you select `<None>`, the system default MoH audio source service parameter (Default Network Hold MoH Audio Source ID) is used for the MoH audio source.

To upload an MOH audio source file that does not appear in the drop-down list, click **Upload File**. In the **Upload File** window, either enter the path of an audio source file or navigate to the file by clicking **Browse**. After you locate the audio source file, click the **Upload File** button to complete the upload. After the audio file gets uploaded, the Upload Result window displays the result of the upload. Click **Close** to close this window.

**Note**  
When you upload a file, the file is uploaded to the 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.

**Note**  
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.

### 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 call-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

Allows line members to log off the hunt list automatically. If an agent does not answer a queuing-enabled hunt pilot call, that agent will be logged off of the hunt group and will not receive additional hunt pilot calls unless he presses the "HLOG" soft key on the phone to log into the hunt pilot.

Line members can log back in using the "HLOG" softkey or PLK.

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

<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 Send PRACK if 1XX contains SDP. 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 Connect Inbound Call before Playing Queuing Announcement checkbox is checked under Device &gt; Device Settings &gt; SIP Profile &gt; Trunk Specific Configuration in Cisco Unified CM Administration. If Connect Inbound Call before Playing Queuing Announcement 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>
## 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.
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.

Note

- 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 58

Configure Call Throttling

• Call Throttling Overview, on page 717
• Call Throttling Configuration Task Flow, on page 718

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 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 Call Throttling, on page 718 Enables Call throttling automatically when your system detects conditions such as heavy call activity, low CPU availability, and disk fragmentation.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Memory Throttling, on page 718 Configures memory throttling for your system.</td>
</tr>
</tbody>
</table>

**Configure Call Throttling**

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 is configured via advanced service parameters. For many deployments, the default settings are sufficient.

⚠️ Caution

We recommend that you not modify call throttling parameters unless advised to do so by customer support.

**Procedure**

1. From Cisco Unified CM Administration, choose System > Service Parameters.
2. From the Server drop-down, choose a server.
3. From the Service drop-down, choose Cisco CallManager.
4. Click Advanced.
5. Under Call Throttling, configure values for the call throttling service parameters. For parameter help descriptions, click the parameter name in the GUI.
   - Code Yellow Entry Latency
   - Code Yellow Exit Latency Calendar
   - Code Yellow Duration
   - Max Events Allowed
   - System Throttle Sample Size
6. Click Save.

**Configure Memory Throttling**

Use this procedure to configure memory throttling for your system.
**Procedure**

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

**Step 2** From the **Server** drop-down, select a Unified Communications Manager server.

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

**Step 4** Click **Advanced**.

**Step 5** Set the **Enable Memory Throttling** parameter to **True**.

**Step 6** Configure values for the additional service parameters in the **Memory Throttling** area. For parameter help, click the parameter name in the GUI.

**Step 7** Click **Save**.
Configure Memory Throttling
Configure Logical Partitioning

- Logical Partitioning Overview, on page 721
- Logical Partitioning Configuration Task Flow, on page 722
- Logical Partitioning Interactions, on page 728
- Logical Partitioning Restrictions, on page 729

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><strong>Step 1</strong></td>
<td>Enable Logical Partitioning, on page 722</td>
<td>Enable Logical Partitioning.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>To Configure Geolocations, on page 722, perform the following subtasks: • Create Geolocations, on page 723 • Assign Geolocations, on page 723 • Set the Default Geolocation, on page 724</td>
<td>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.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Logical Partitioning Default Policy, on page 724</td>
<td>Purpose</td>
</tr>
<tr>
<td>--------</td>
<td>-------------------------------------------------------------</td>
<td>---------</td>
</tr>
<tr>
<td></td>
<td></td>
<td>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.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Devices to Avoid Logical Partitioning Checks, on page 724</td>
<td>Purpose</td>
</tr>
<tr>
<td></td>
<td></td>
<td>You can specifically exempt devices and device pools from the partitioning checks.</td>
</tr>
<tr>
<td>Step 5</td>
<td>To Configure Geolocation Filters, on page 725, perform the following subtasks:</td>
<td>Purpose</td>
</tr>
<tr>
<td></td>
<td>• Create Geolocation Filter Rules, on page 725</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>• Assign Geolocation Filters, on page 726</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Set the Default Geolocation Filter, on page 726</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>Define a Set of Logical Partitioning Policy Records, on page 727</td>
<td>Purpose</td>
</tr>
<tr>
<td></td>
<td></td>
<td>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.</td>
</tr>
<tr>
<td>Step 7</td>
<td>(Optional) Enable Location Conveyance, on page 727</td>
<td>Purpose</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Configure location conveyance if you want to communicate geolocation information about devices across clusters.</td>
</tr>
</tbody>
</table>

**Enable Logical Partitioning**

Use this procedure to turn on the Logical Partitioning feature.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose System &gt; Enterprise Parameters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>For the <strong>Enable Logical Partitioning</strong> enterprise parameter, choose True from the drop-down list.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

**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>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Create Geolocations, on page 723</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>Step 2 Assign Geolocations, on page 723</td>
<td>Assign a geolocation to a device or device pool.</td>
</tr>
<tr>
<td>Step 3 Set the Default Geolocation, on page 724</td>
<td>Specify a default geolocation for all devices and device pools in this cluster.</td>
</tr>
</tbody>
</table>

**Assign Geolocations**

Assign a geolocation to a device or device pool.

**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.
• 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>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></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>Create Geolocation Filter Rules, on page 725</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></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>
<tr>
<td>Assign Geolocation Filters, on page 726</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Set the Default Geolocation Filter, on page 726</td>
</tr>
</tbody>
</table>
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.

Assign Geolocation Filters

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

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.
Step 6: Click **Save**.

---

## Logical Partitioning Interactions

**Table 70: 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. |
| Meet-Me Conference | Logical partitioning handling does not take place in the following circumstances:  
• When all participants are VoIP phones, handling does not occur.  
• When geolocation or geolocation filter does not associate with a device, no policy check takes place for that device. |
Interaction Feature

Logical partitioning handling does not take place in the following circumstances:

- When both the calling party and called party devices are VoIP phones, handling does not occur.
- All devices must associate with both a geolocation and geolocation filter. If any device does not associate with both geolocation and geolocation filter, handling does not occur.

Route Lists and Hunt Pilots

 Logical partitioning handling does not take place in the following circumstances:

- When both the caller and the callee devices are VoIP phones, no handling occurs.
- When geolocation or geolocation filter does not associate with any device, no handling occurs.

Shared Line

 Logical partitioning handling does not take place in the following circumstances:

Cisco Unified Communications Manager does not communicate geolocation information over a H.225 gatekeeper-controlled trunk.

### Logical Partitioning Restrictions

#### Table 71: 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.</td>
</tr>
<tr>
<td></td>
<td>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>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>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</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>
CHAPTER 60

Configure Location Awareness

- Location Awareness Overview, on page 731
- Location Awareness Prerequisites, on page 733
- Location Awareness Configuration Task Flow, on page 733

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 Responder checks for updates to the existing associations. As a result, even if a mobile caller places an
emergency call while in a roaming situation, Cisco Emergency Responder can quickly determine the physical location of the caller and send emergency services to the appropriate building, floor, or cube.

**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
- Cisco Jabber clients—supported as of 12.5(1)SU1

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>Step 1</td>
<td>Start Services for Wireless Infrastructure Synchronization, on page 734</td>
<td>In Cisco Unified Serviceability, start services that support the Location Awareness feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Wireless Access Point Controller, on page 734</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>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
</tbody>
</table>
| **Step 3** Insert Infrastructure Devices, on page 735 | Optional. If you want to add your wireless infrastructure from Cisco Prime Infrastructure, or if you are using a third-party wireless LAN controller, use Bulk Administration to update the database from a CSV file.

**Note** This method does not allow you to set up automatic updates. |

| **Step 4** Deactivate Infrastructure Device from Tracking, on page 736 | Optional. If your synchronization includes access points that you do not want to track (for example, if the synchronization pulls in access points from a lab), you can deactivate the access point and Cisco Unified Communications Manager will not track updates to the access point. |

### 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**

1. **Step 1** Log in to Cisco Unified Serviceability and choose Tools > Service Activation.
2. **Step 2** From the Server drop-down list, select the publisher node.
3. **Step 3** Make sure that the following services are checked:
   - Cisco CallManager
   - Cisco AXL Web Service
   - Cisco Wireless Controller Synchronization Service
4. **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.
5. **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 box, 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 the system Online Help.
  c) Click the Test SNMP Settings button 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.

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
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 to 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 741</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 743</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 In Cisco Unified Communications Manager 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 Bandwidth Pool 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 the Related Topics section for information about the service parameters and their configuration options.

Step 6 Click Save.

Related Topics
Flexible DSCP Marking and Video Promotion Service Parameters, on page 742

Flexible DSCP Marking and Video Promotion Service Parameters

**Note**
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.

<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>
</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 743</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 744</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

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

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 745
Flexible DSCP Marking and Video Promotion Interactions

Table 73: 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 74: 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>
Separate Calling Party Number and Billing Number in SIP

• External Presentation Name and Number Overview, on page 747
• Call Processing, on page 748
• Directory Number Overview, on page 750
• SIP Profile Overview, on page 754
• SIP Trunk Overview, on page 756
• Intercluster SME Call Flows, on page 760

External Presentation Name and Number Overview

Cisco Unified Communications Manager Administration can be configured to contain separate calling party and presentation number.

In previous releases, Cisco Unified Communications Manager cannot be configured on a per line basis to have a different number sent to the PSTN in the FROM and PAID header. If a group of users is configured to present the same Calling Line Identification Number to PSTN users, which is a nongeographic E.164 number and cannot be used for billing. Therefore, the users actual DDI must be sent in a different field than the presentation number. With this release, Cisco Unified Communications Manager supports External Presentation Name and Number that is different from existing Identification Name and Number. The configured Presentation Name and Number are for display purpose on the following devices:

• SIP
• SCCP
• Single Number Reach Destination (SNRD)
• CTIRD
• SparkRD

Configuration Overview

You can configure the external presentation name and number feature on the following pages:

• Directory Number Configuration
• SIP Profile Configuration
• Trunk Configuration

---

**Note**

• When you configure the External Presentation Information on the SIP Profile Configuration page, the value of **External Presentation Number** and **External Presentation Name** on the SIP Profile Configuration page is used, overriding the settings configured on the Directory Number page.

• When you configure the Presentation Information on the Trunk Configuration page, the value of **Presentation Number** and **Presentation Name** on the Trunk Configuration page is used, overriding the settings configured on the SIP Profile Configuration and Directory Number Configuration pages.

---

**Call Processing**

This section describes the incoming and outgoing call behavior when you configure the external presentation name and number feature.

**Incoming Call Process**

Cisco Unified Communications Manager looks for FROM and PAID header information when there is a call initiated from the PSTN network. The FROM header contains the external presentation name and number (if configured). However, this is not the real identity of a user, it is used only for display purpose. The PAID header contains the identity (original DN or DDI) of the user.

If FROM and PAID headers have different numbers and **Enable External Presentation Name and Number** option is enabled in the SIP Profile Configuration page and **Display External Presentation Name and Number** service parameter value is set to **True**, then Cisco Unified Communications Manager displays the FROM header information (configured external presentation name and number) on the called device. Similarly, if an option is disabled, Cisco Unified Communications Manager displays PAID header information (user's original DN or DDI) on the called device.

---

**Note**

• By default, **Enable External Presentation Name and Number** field is unchecked.

• Default value of a service parameter **Display External Presentation Name and Number** is False.

---

**Invite Received from the PSTN Network**

From: "Customer Care" <sip:1800000@example.com>;  
To: <sip:someone@example.com>  
P-Asserted-Identity: "Your personal adviser <sip:user1@example.com>  
Remote-Party-ID: "Your personal adviser <sip:user1@example.com>  

In the preceding example, FROM header contains a number different from the PAID header. If you check **Enable External Presentation Name and Number** check box and set **Display External**
Presentation Name and Number value to True, Cisco Unified Communications Manager displays Customer Care / 1800000 on the called device.

If you uncheck the Enable External Presentation Name and Number check box or set the Display External Presentation Name and Number to False, then Cisco Unified Communications Manager displays Your personal adviser / user1@example.com on the called device.

Outgoing Call Process

Let us assume, a user configured with External Presentation Name and Number initiates a call to a PSTN network through the SIP trunk with Enable External Presentation Name and Number configured in its SIP Profile. Then, Cisco Unified Communications Manager sends the configured External Presentation Information in the FROM header of the outgoing SIP message and displays on the called device.

If Enable External Presentation Name and Number option is disabled or External Presentation Number and External Presentation Name fields are not configured, the Cisco Unified Communications Manager sends the directory number information in the FROM and PAID headers and displays on the called device.

External Presentation Number Mask Operation

Cisco Unified Communications Manager allows you to mask the external presentation number, to be displayed on the called device. You can mask the presentation number on the Directory Number Configuration, SIP Profile Configuration, and Trunk Configuration pages.

When you enter the digits in the External Presentation Number field with trailing X, the value of X is replaced with the directory number information starting from right to left.

Mask Operation on Directory Number Configuration

If you mask External Presentation Number as 180011XXXX on Directory Number Configuration page for a Directory Number 5551234, then Cisco Unified Communications Manager displays the presentation number as 1800111234 on the called device.

Mask Operation on SIP Profile Configuration

Let us assume External Presentation Number on Directory Number page is 180011XXXX and if you mask External Presentation Number on SIP Profile Configuration page as 180022XXXX for a Directory Number 5551234, then Cisco Unified Communications Manager displays the presentation number as 1800221234 on the called device.

Mask Operation on Trunk Configuration

Let us assume External Presentation Number on Directory Number page is 180011XXXX and on SIP Profile Configuration page is 180022XXXX. If you mask Presentation Number on Trunk Configuration page as 180033XXXX for a Directory Number 5551234, then Cisco Unified Communications Manager displays the presentation number as 1800331234 on the called device.
Directory Number Overview

In Cisco Unified Communications Manager Administration, use the Call Routing > Directory Number menu path to configure Directory Numbers (DNs). Using Cisco Unified Communications Manager Administration, you can configure and modify the DNs that are assigned to specific phones.

A new section External Presentation Information is added on the Directory Number Configuration page. The administrator can now configure the presentation name and number of their choice to display on the supported devices for external calls. If an administrator does not want to show users' identity, they have a privilege to display configured External Presentation Number and External Presentation Name as Anonymous on the called party device.

Directory Number Configuration Tasks

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | Add a new end user using one of the following methods:  
• Import an End User from LDAP, on page 750  
• Add an End User Manually, on page 751 | If your system is synchronized with a company LDAP directory, you can import the new end user directly from LDAP. Otherwise, you can add and configure the end user manually. |
| **Step 2** | Assign a phone to new or existing end user by performing one of the following tasks:  
• Add New Phone for End User, on page 752  
• Move an Existing Phone to a End User, on page 753 | You can use the 'Add New Phone' procedure to configure a new phone for the end user using settings from a universal device template. You can also use the 'Move' procedure to assign an existing phone already configured or pre-configured. |
| **Step 3** | Configure External Presentation Information on DN, on page 753 | To configure the external presentation number and external presentation name for DNs that are assigned to specific phones. |

Import an End User from LDAP

Perform the following procedure to manually import a new end user from a company LDAP directory. If your LDAP synchronization configuration includes a feature group template with a user profile that includes universal line and device templates and a DN pool, the import process automatically configures the end user and primary extension.

You cannot add new configurations (for example, adding a feature group template) into an LDAP directory sync after the initial sync has occurred. If you want to edit an existing LDAP sync, you must either use Bulk Administration, or configure a new LDAP sync.

Note

You cannot add new configurations (for example, adding a feature group template) into an LDAP directory sync after the initial sync has occurred. If you want to edit an existing LDAP sync, you must either use Bulk Administration, or configure a new LDAP sync.
Before you begin

Before you begin this procedure make sure that you have already synchronized Cisco Unified Communications Manager with a company LDAP directory. The LDAP synchronization must include a feature group template with universal line and device templates.

Procedure

Step 1
In Cisco Unified CM Administration, choose System > LDAP > LDAP Directory.

Step 2
Click Find and select the LDAP directory to which the user is added.

Step 3
Click Perform Full Sync. Cisco Unified Communications Manager synchronizes with the external LDAP directory. Any new end users in the LDAP directory are imported into the Cisco Unified Communications Manager database.

What to do next

If the user is enabled for self-provisioning, the end user can use the Self-Provisioning Interactive Voice Response (IVR) to provision a new phone. Otherwise, perform one of the following tasks to assign a phone to the end user:

• Add New Phone for End User, on page 752
• Move an Existing Phone to a End User, on page 753

Add an End User Manually

Perform the following procedure to add new end user and configure them with an access control group and a primary line extension.

Before you begin

Verify that you have a user profile configured that includes a universal line template. If you need to configure a new extension, Cisco Unified Communications Manager uses the settings from the universal line template to configure the primary extension.

Procedure

Step 1
In Cisco Unified CM Administration, choose User Management > User/Phone Add > Quick User/Phone Add.

Step 2
Enter the User ID and Last Name.

Step 3
From the Feature Group Template drop-down list, select a feature group template.

Step 4
Click Save.

Step 5
From the User Profile drop-down list, verify that the selected user profile includes a universal line template.

Step 6
From the Access Control Group Membership section, click the + icon.

Step 7
From the User is a member of drop-down list, select an access control group.

Step 8
Under Primary Extension, click the + icon.
Step 9  From the Extension drop-down list, select a DN that displays as (available).

Step 10 If all line extensions display as (used), perform the following steps:
   a) Click the New... button.
      The Add New Extension popup displays.
   b) In the Directory Number field, enter a new line extension.
   c) From the Line Template drop-down list box, select a universal line template.
   d) Click OK.
      Cisco Unified Communications Manager configures the directory number with the settings from the
      universal line template.

Step 11 (Optional) Complete any additional fields in the Quick User/Phone Add Configuration window.

Step 12 Click Save.

What to do next

Perform one of the following procedures to assign a phone to this end user:

   • Add New Phone for End User, on page 752
   • Move an Existing Phone to a End User, on page 753

Add New Phone for End User

Perform the following procedure to add a new phone for a new or existing end user. Make sure that the user
profile for the end user includes a universal device template. Cisco Unified Communications Manager uses
the universal device template settings to configure the phone.

Before you begin

Perform one of the following procedures to add an end user:

   • Add an End User Manually, on page 751
   • Import an End User from LDAP, on page 750

Procedure

Step 1 In Cisco Unified CM Administration, choose User Management > User/Phone Add > Quick/User Phone
Add.
Step 2 Click Find and select the end user for whom you want to add a new phone.
Step 3 Click the Manage Devices button.
   The Manage Devices window appears.
Step 4 Click Add New Phone.
   The Add Phone to User popup displays.
Step 5 From the Product Type drop-down list, select the phone model.
Step 6 From the Device Protocol drop-down select SIP or SCCP as the protocol.
Step 7 In the Device Name text box, enter the device MAC address.
Step 8: From the **Universal Device Template** drop-down list, select a universal device template.

Step 9: If the phone supports expansion modules, enter the number of expansion modules that you want to deploy.

Step 10: If you want to use Extension Mobility to access the phone, check the **In Extension Mobility** check box.

Step 11: Click **Add Phone**. The Add New Phone popup closes. Cisco Unified Communications Manager adds the phone to the user and uses the universal device template to configure the phone.

Step 12: If you want to make additional edits to the phone configuration, click the corresponding Pencil icon to open the phone in the **Phone Configuration** window.

---

## Move an Existing Phone to a End User

Perform this procedure to move an existing phone to a new or existing end user.

**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 <strong>User Management &gt; User/Phone Add &gt; Quick/User Phone Add</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong> and select the user to whom you want to move an existing phone.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click the <strong>Manage Devices</strong> button.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click the <strong>Find a Phone to Move To This User</strong> button.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Select the phone that you want to move to this user.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click <strong>Move Selected</strong>.</td>
</tr>
</tbody>
</table>

---

## Configure External Presentation Information on DN

Perform the following procedure to configure the external presentation information for DNs that are assigned to specific phones.

**Before you begin**

- Check the **Enable External Presentation Name and Number** check box on the SIP Profile Configuration page.
- Perform one of the following procedures to add an end user:
  - **Add an End User Manually**, on page 751
  - **Import an End User from LDAP**, on page 750
- Assign a phone to a new or existing end user by performing one of the following tasks:
  - **Add New Phone for End User**, on page 752
  - **Move an Existing Phone to a End User**, on page 753
**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose **Call Routing > Directory Number**.

**Step 2**  From the **Find and List Directory Numbers** page, perform one of the following steps:
- To update a DN, click **Find** and select the Directory Number for which you want to display unique identity.
- To create a new Directory Number, click **Add New**.

**Step 3**  In the **External Presentation Information** section, enter the name and number that you want to display on the called device.

**Note**  
- **External Presentation Number** field accepts up to 32 digits and can contain the following characters: [0-9, X, *, #, +].
- **External Presentation Name** field accepts up to 50 characters.

**Step 4**  (Optional), if you want to show configured **External Presentation Number** and **External Presentation Name** as anonymous, check the **Anonymous External Presentation** check box.

**Note**  
- By default, the **Anonymous External Presentation** field is unchecked.
- If you check the **Anonymous External Presentation** field:
  - **External Presentation Number** and **External Presentation Name** fields are noneditable.
  - Also, the entries from these fields disappear.

**Step 5**  Complete the remaining fields on the **Directory Number Configuration** page. For help with the fields and their settings, refer to the online help.

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

---

**SIP Profile Overview**

A SIP profile is a template that comprises common SIP settings. You must assign a SIP profile for every SIP trunk and SIP device in your network. When you configure a SIP profile and then assign that profile to a SIP trunk, or a SIP device, the system applies the configured SIP settings to that trunk or device.

**SIP Profile Configuration Tasks**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure SIP Profiles, on page 755</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure External Presentation Information on SIP Profile, on page 755</td>
</tr>
</tbody>
</table>
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:
- To edit an existing profile, click Find and select the SIP profile.
- To create a new profile, click Add New.

**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 box.

**Step 5**
If you want to assign a normalization or transparency script to resolve SIP interoperability issues, from the Normalization Script drop-down, 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 help with the fields and their settings, refer to the online help.

**Step 8**
Click Save.

What to do next

In order for a trunk or device to be able to use a SIP profile, you must associate the profile to a trunk or device in the Trunk Configuration or Phone Configuration windows.

Configure External Presentation Information on SIP Profile

Use this procedure to configure the separate external presentation name and number on SIP Profile Configuration page.

**Before you begin**

- Check the Enable External Presentation Name and Number check box on the SIP Profile Configuration page.
- Set Display External Presentation Name and Number parameter value to True under Clusterwide Parameters (Device-Phone) section on Service Parameter Configuration page.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile.

**Step 2**
Perform one of the following steps:
• To edit an existing profile, click **Find** and select the SIP profile.
• To create a new profile, click **Add New**.

**Step 3**
In the **External Presentation Information** section, enter the name and number that you want to display on the called device.

**Note**
- **External Presentation Number** field accepts up to 32 digits and can contain the following characters: [0-9, X, *, #, 
+].
- **External Presentation Name** field accepts a maximum of 50 characters.

**Step 4**
(Optional), if you want to show configured **External Presentation Number** and **External Presentation Name** as anonymous, check the **Anonymous External Presentation** check box.

**Note**
- By default, the **Anonymous External Presentation** field is unchecked.
- If you check the **Anonymous External Presentation** field:
  - **External Presentation Number** and **External Presentation Name** fields are noneditable. Also, the entries from these fields disappear.

**Step 5**
Complete the remaining fields in the **SIP Profile Configuration** page. For more information on the fields and their configuration options, see the system Online Help.

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

---

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

SIP Trunk supports the separate presentation name and number that is different from existing Caller ID DN and Caller Name. A new check box **Anonymous Presentation** is provided to show the configured presentation name and number as Anonymous on the called device.

**Trunk Configuration Tasks**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure SIP Trunk Security Profile, on page 757</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.</td>
</tr>
</tbody>
</table>
If you don't configure SIP trunk security profiles, by default, Cisco Unified Communications Manager applies a nonsecure sip trunk security profile.

**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 box, 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.
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:

- **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.

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 box, choose SIP Trunk.
Step 4  From the Protocol Type drop-down list box, 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.
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 (+) button.
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, 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 box, select the script that you want to assign.
Step 12 Configure any additional fields in the Trunk Configuration window. For help with the fields and their settings, refer to the online help.
Step 13 Click Save.

Configure Presentation Information on SIP Trunks

Use this procedure to configure the presentation name and number on SIP Trunk page.

Before you begin
   • Check the Enable External Presentation Name and Number check box on the SIP Profile Configuration page.
   • Configure SIP Trunks, on page 758
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 box, choose SIP Trunk.

Step 4
From the Trunk Service Type drop-down list box, choose the type of SIP trunk that you want to configure:

- **None (Default):** The trunk will not be used for Call Control Discovery, Extension Mobility Cross-Cluster, Intercompany Media Engine, or IP Multimedia System Service Control.
- **Call Control Discovery:** The trunk supports the Call Control Discovery feature.
- **Extension Mobility Cross Cluster:** The trunk supports Extension Mobility Cross Cluster.
- **Cisco Intercompany Media Engine:** The trunk supports the Intercompany Media Engine (IME). Make sure that the IME server is installed before you configure this type of trunk.
- **IP Multimedia System Service Control:** Choose this option to enable the trunk with support for IP Multimedia System Service Control.

Step 5
Click Next.

Step 6
In the Presentation Information section, enter the name and number that you want to display on the called device.

**Note**

- **Presentation Number** field accepts up to 32 digits and can contain the following characters: [0-9, X, *, #, \, +].
- **Presentation Name** field accepts a maximum of 50 characters.

Step 7
(Optional), if you want to show the presentation name and number as anonymous, check the Anonymous Presentation check box.

**Note**

- By default, the Anonymous Presentation field is unchecked.
- If you check the Anonymous External Presentation field:
  - **Presentation Number** and **Presentation Name** fields are noneditable. Also, the entries from these fields disappear.

Step 8
(Optional), check the **Send Presentation Name and Number only in the FROM header and not in the other identity headers** check box, if you want to send presentation information that is configured on the SIP Trunk only to FROM header.

Step 9
Configure any additional fields in the Trunk Configuration window. For help with the fields and their settings, refer to the online help.

Step 10
Click Save.

---

**Intercluster SME Call Flows**

The Cisco Unified Communications Manager Session Management Edition software is same as the Cisco Unified Communications Manager used mainly for a call routing between clusters or various devices. With this release, Cisco Unified Communications Manager supports the intercluster SME calls.
Incoming Calls

Let us assume a user from a PSTN network initiates a call with **Enable External Presentation Name and Number** enabled in its SIP profile. If **Display External Presentation Name and Number** service parameter is set to **True**, then Cisco Unified Communications Manager sends the presentation number information to the X-Cisco-Presentation header and displays on the called device. The FROM and PAID headers contain the identity of the user that is the user's DN or DDI.

If **Display External Presentation Name and Number** service parameter is set to **False**, then Cisco Unified Communications Manager sends the presentation number information to the X-Cisco-Presentation header. The FROM and PAID headers contain the user's DN or DDI and displays on the called device.

Outgoing Calls

A user who is configured with **External Presentation Name** and **External Presentation Number** initiates a call to a PSTN network through intercluster SIP trunks. If **Enable External Presentation Name and Number** check box is disabled in its SIP profile, then, Cisco Unified Communications Manager sends the original directory number information in the FROM and PAID headers and displays on the called device and configured External Presentation Information in the X-Cisco-Presentation header. Similarly, if **Enable External Presentation Name and Number** check box is enabled in its SIP profile, Cisco Unified Communications Manager sends the configured External Presentation Information in the FROM header and displays on the called device and original Directory Number in the PAID header.
SIP OAuth Mode

- SIP OAuth Mode Overview, on page 763
- SIP OAuth Mode Prerequisites, on page 763
- SIP OAuth Mode Configuration Task Flow, on page 764

SIP OAuth Mode Overview

Secure registrations to Unified Communications Manager involves a process of updating CTL files, setting up a mutual certificate trust store and so on. If a Cisco Jabber device is switching between on-premises and off-premises, it is difficult to update LSCs and renew Certificate Authority Proxy Function (CAPF) enrollment each time when a secure registration is completed.

SIP OAuth mode allows you to use OAuth refresh tokens for Cisco Jabber authentication in secure environments. Supporting OAuth on the Unified Communications Manager SIP line allows secure signalling and media without CAPF. OAuth token validation during SIP registration is completed when OAuth based authorization is enabled on Unified Communication Manager cluster and Cisco Jabber endpoints.

OAuth support for SIP registrations is extended only for Cisco Jabber devices from Cisco Unified Communications Manager 12.5 release onwards.

The following are the Phone Security Profile Types that can be configured for OAuth. Currently, this is supported only for Cisco Jabber.

- Cisco Dual Mode For iPhone (TCT device)
- Cisco Dual Mode For Android (BOT device)
- Cisco Unified Client Service Framework (CSF device)
- Cisco Jabber for Tablet (TAB device)
- Universal Device Template

SIP OAuth Mode Prerequisites

This feature assumes that you have already completed the following:

- Ensure that Mobile and Remote Access is configured and the connection is established between Unified Communication Manager and Expressway.
• Ensure that Unified Communications Manager is registered to a Smart or Virtual account with allow export-controlled functionality.

## SIP OAuth Mode Configuration Task Flow

Complete the following tasks to configure SIP OAuth for your system.

### Procedure

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<thead>
<tr>
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<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Refresh Logins, on page 764</td>
<td>Enable oauth with refresh login flow on Unified Communications Manager to register the device via SIP OAuth.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure OAuth Ports, on page 765</td>
<td>Assign the ports for OAuth for each node that has OAuth registration.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure OAuth Connection to Expressway-C, on page 765</td>
<td>Configure a mutually authenticated TLS connection to Expressway-C.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Enable SIP OAuth Mode, on page 765</td>
<td>Enable OAuth services using a CLI command on the publisher node.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Restart Cisco CallManager Service, on page 766</td>
<td>Restart this service on all nodes that have OAuth registrations.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Configure OAuth Support in Security Profile, on page 766</td>
<td>Configure OAuth support within a Phone Security Profile if you are deploying encryption for the endpoints.</td>
</tr>
</tbody>
</table>

### Configure Refresh Logins

Use this procedure to configure Refresh Logins with OAuth access tokens and refresh tokens for Cisco Jabber clients.

**Procedure**

1. From Cisco Unified CM Administration, choose **System > Enterprise Parameters**.
2. Under **SSO and OAuth Configuration**, set the **OAuth with Refresh Login Flow** parameter to **Enabled**.
3. (Optional) Set any other parameters in the **SSO and OAuth Configuration** section. For parameter descriptions, click on the parameter name.
4. Click **Save**.
Configure OAuth Ports

Use this procedure to assign the ports that are used for SIP OAuth.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Cisco Unified CM.
Step 2 Do the following for each server that uses SIP OAuth.
Step 3 Select the server.
Step 4 Under Cisco Unified Communications Manager TCP Port Settings, set the port values for the following fields:

- SIP Phone OAuth Port
  Default value is 5090. Acceptable configurable range is 1024 to 49151.

- SIP Mobile and Remote Access Port
  Default value is 5091. Acceptable configurable range is 1024 to 49151.

Step 5 Click Save.
Step 6 Repeat this procedure for each server that uses SIP OAuth.

Configure OAuth Connection to Expressway-C

Use this procedure to add the Expressway-C connection to Cisco Unified Communications Manager. You need this configuration for devices in Mobile and Remote Access mode with SIP OAuth.

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose Device > Expressway-C.
Step 2 Click Add New.
Step 3 Enter an IP Address, Hostname or fully qualified domain name for the Expressway-C.
Step 4 Enter a Description.
Step 5 Enter the X.509 Subject Name/Subject Alternate Name of the Expressway-C from the Expressway-C certificate.
Step 6 Click Save.

Enable SIP OAuth Mode

Use the Command Line Interface to enable SIP OAuth mode. Enabling this feature on the publisher node also enables the feature on all cluster nodes.
Procedure

Step 1 On the Unified Communications Manager publisher node, log in to the Command Line Interface.
Step 2 Run the `utils sip-oauth enable` CLI command.

Restart Cisco CallManager Service

After enabling SIP OAuth through CLI, restart the Cisco CallManager service on all nodes where endpoints register through SIP OAuth.

Procedure

Step 1 From Cisco Unified Serviceability, choose Tools > Control Center – Network Services.
Step 2 From the Server drop-down, select the server.
Step 3 Check the Cisco CallManager service and click Restart.

Configure OAuth Support in Security Profile

Use this procedure to configure OAuth authentication if you are deploying encrypted endpoints. This procedure is required only if you have set the Device Security Mode within that phone’s Phone Security Profile to Encrypted.

Procedure

Step 1 From Cisco Unified CM Administration, choose System > Phone Security Profile.
Step 2 Click Find and select the security profile that is used by the phone.
Step 3 Ensure that the Device Security Mode is Encrypted and the Transport Type is TLS.
Step 4 Check the Enable OAuth Authentication check box.
Step 5 Click Save.

Note When SIP OAuth Mode is enabled, Enable Digest Authentication and TFTP Encrypted Config options are not supported.
PART XIV

QoS Management

- Configure QoS with APIC-EM Controller, on page 769
- Configure AS-SIP Endpoints, on page 775
- Configure Multilevel Precedence and Preemption, on page 787
Configure QoS with APIC-EM Controller

- APIC-EM Controller Overview, on page 769
- APIC-EM Controller Prerequisites, on page 770
- APIC-EM Controller Configuration Task Flow, on page 770

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 741.
- 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.

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
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<td>Configure Unified CM on the APIC-EM Controller.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Upload APIC-EM Controller Certificate, on page 771</td>
<td>Upload the APIC-EM certificate into Cisco Unified OS Administration.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure HTTPS Connection to APIC-EM Controller, on page 772</td>
<td>Configure an HTTPS Profile that points to the APIC-EM service.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Enable External QoS Service for System, on page 772</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. <strong>Note</strong> 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 5</td>
<td>Configure External QoS Service at SIP Profile Level, on page 772</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 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 6</strong></td>
<td>Assign SIP Profile to Phones, on page 773</td>
</tr>
</tbody>
</table>

**Configure the APIC-EM Controller**

You can use the SIP Profile setting to configure which devices and device types you want the APIC-EM to manage media flows.

Associate the external QoS-enabled SIP Profile to a phone.

**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**.

Upload APIC-EM Controller Certificate

Use this procedure to upload the APIC-EM controller certificate into 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></td>
<td>From Cisco Unified OS Administration, choose <strong>Security &gt; Certificate Management</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click <strong>Upload Certificate/Certificate Chain</strong>. The <strong>Upload Certificate/Certificate Chain</strong> popup window appears.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>From the <strong>Certificate Purpose</strong> drop-down list, choose <strong>CallManager-trust</strong>.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Enter a <strong>Description</strong> for the certificate.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click <strong>Browse</strong> to search for, and select, the certificate.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Click <strong>Upload</strong>.</td>
</tr>
</tbody>
</table>
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.

**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.

Enable External QoS Service for System

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.

**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.

Configure External QoS Service at SIP Profile Level

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.

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 Cisco Unified Communications Manager.

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.
Assign SIP Profile to Phones
Configure AS-SIP Endpoints

• AS-SIP Overview, on page 775
• AS-SIP Prerequisites, on page 777
• AS-SIP Endpoint Configuration Task Flow, on page 778

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="GBauADsa2qoWr6k9y3hGGVDAqnLfoLkS",uri ="sip:172.18.197.224",
algorithm=MD5,response="126c0643a4923359ab59d4f53494552e"
```

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 75: Comparison of the Configuration Differences Between Cisco IP Phones and Third-Party Phones**

<table>
<thead>
<tr>
<th>Phone Running AS-SIP</th>
<th>Integrated with Centralized TFTP</th>
<th>Sends MAC Address</th>
<th>Downloads Softkey File</th>
<th>Downloads Dial Plan File</th>
<th>Supports Unified Communications Manager Failover and Fallback</th>
<th>Supports Reset and Restart</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 (For more information, see "Configure SIP Profile" topic in System Configuration Guide for Cisco Unified Communications Manager).

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 more information, see "Smart Software Licensing" chapter from System Configuration Guide for Cisco Unified Communications Manager.
# AS-SIP Endpoint Configuration Task Flow

Complete the following tasks to configure an AS-SIP endpoint.

<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 779</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 779</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 779</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 780</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 781</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 781</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 782</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 783</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 783</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 784</td>
<td>Configure additional AS-SIP features such as MLPP, TLS, V.150 and IPv6.</td>
</tr>
</tbody>
</table>
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.

Procedure

Step 1
From Cisco Unified CM Administration, choose User Management > End User.

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

Step 3
Make sure the following mandatory fields are completed:
- User ID
- Last Name

Step 4
In the Digest Credentials field, enter a password. End users must authenticate themselves via this password when using the endpoint.

Step 5
Complete any remaining fields. For help with the fields and their settings, see the online help.

Step 6
Click Save.

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.
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 box, 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 help with the fields and their settings, refer to the 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.

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Security > Phone Security Profile.
Step 2  Perform one of the following steps:
   - Click Add New to create a new phone security profile.
   - Click Find to edit an existing profile.
Step 3  For new profiles, select an option from the Phone Security Profile drop-down, choose the phone model Third-party AS-SIP Endpoint and click Next.
   - For Cisco IP phones, select the phone model and click Next.
   - For third-party AS-SIP endpoints, select Third-party AS-SIP Endpoint and click Next.
Step 4  For the protocol, select SIP and click Next.
Step 5  Enter a Name and Description for the protocol.
Step 6  Assign the Device Security Mode, to one of the following settings:
   - Authenticated—Cisco Unified Communications Manager uses TLS signaling, providing integrity and authentication for the phone.
   - Encrypted—Cisco Unified Communications Manager uses TLS signaling, providing integrity and authentication for the phone. In addition, SRTP encrypts the media streams.
Step 7  Check the Enable Digest Authentication check box.
Step 8  Configure the remaining fields in the Phone Security Profile Configuration window. For help with the fields and their settings, see the online help.
Step 9  Click Save.

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, 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 help with the fields and their settings, see the online help:

- Device Trust Mode—For third-party AS-SIP endpoints only. Select Trusted or Not Trusted.
- MAC Address
- Device Pool
- 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified CM Administration, choose <strong>User Management &gt; End User</strong>.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click <strong>Find</strong> and select the user whom you want to associate to the device.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>In the <strong>Device Information</strong> section, click <strong>Device Association</strong>. The User Device Association window appears.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Click <strong>Find</strong> to view a list of available devices.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Select the device that you want to associate, and click <strong>Save Selected/Changes</strong>.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>From <strong>Related Links</strong>, choose <strong>Back to User</strong>, and click <strong>Go</strong>. The <strong>End User Configuration</strong> window appears, and the associated device that you chose appears in the <strong>Controlled Devices</strong> pane.</td>
</tr>
</tbody>
</table>
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**

<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; Security &gt; SIP Trunk Security Profile.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Enter a Name for the security profile.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Device Security Mode drop-down, choose Authenticated or Encrypted.</td>
</tr>
<tr>
<td>Step 5</td>
<td>The Incoming Transport Type and Outgoing Transport Type fields change to TLS automatically.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Check the Enable Digest Authentication check box.</td>
</tr>
<tr>
<td>Step 7</td>
<td>If you are deploying V.150, configure a value for the SIP V.150 Outbound SDP Offer Filtering drop-down.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Complete the remaining fields in the SIP Trunk Security Profile Configuration window. For help with the fields and their settings, see the online help.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Configure SIP Trunk for AS-SIP

Use this procedure to set up a SIP trunk that supports AS-SIP.

**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>Do either of the following:</td>
</tr>
<tr>
<td></td>
<td>• Click Find and select an existing trunk.</td>
</tr>
<tr>
<td></td>
<td>• Click Add New to create a new trunk.</td>
</tr>
<tr>
<td>Step 3</td>
<td>For new trunks, from the Trunk Type drop-down list box, select SIP Trunk.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the Trunk Service Type drop-down list box, select None (Default) and click Next.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Enter a Device Name for the trunk.</td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Device Pool drop-down list box, select a device pool.</td>
</tr>
<tr>
<td>Step 7</td>
<td>In the Destination Address field, enter the address of the server to which you are connecting the trunk.</td>
</tr>
<tr>
<td>Step 8</td>
<td>From the SIP Trunk Security Profile drop-down, select the profile that you created for AS-SIP.</td>
</tr>
<tr>
<td>Step 9</td>
<td>From the SIP Profile drop-down, select the SIP Profile that you set up for AS-SIP.</td>
</tr>
<tr>
<td>Step 10</td>
<td>Complete any remaining fields in the Trunk Configuration window. For help with the fields and their settings, see the online help.</td>
</tr>
<tr>
<td>Step 11</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>
# 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>
</table>
| Early Offer    | SIP Early Offer allows your endpoints to negotiate media during the INVITE request and the 200OK response. There are two modes for Early Offer:  
• Best Effort Early Offer (no MTP Inserted)  
• Mandatory Early Offer (insert MTP if needed)  
Configure Early Offer support via the fields in the following configuration windows. Refer to the online help for detailed field descriptions:  
**SIP Profile Configuration** window  
• Early Offer support for voice and video calls—Configure this field to enable Early Offer support on a SIP trunk  
• SDP Session-level Bandwidth Modifier for Early Offer and Re-invite  
• Send send-receive SDP in mid-call INVITE  
**Phone Configuration** window (only if the Third Pary AS-SIP Endpoint device type is used)  
• Early Offer support for voice and video calls - check this check box to enable early offer support |
| Conference Factory | Specify the URI that an IMS client uses to set up a conference:  
1. From Cisco Unified CM Administration, choose **System > Service Parameters**.  
2. From the **Server** drop-down select your Cisco Unified Communications Manager server.  
3. From the **Service**, select **Cisco CallManager**.  
4. Under **Clusterwide Paramters (Feature - Conference)** assign an IMS Conference Factory URI.  
5. Click **Save**. |
<table>
<thead>
<tr>
<th>AS-SIP Feature</th>
<th>Configuration Description</th>
</tr>
</thead>
<tbody>
<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 <a href="#">DSCP Settings Configuration Task Flow, on page 741</a>.</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. For more information to configure IPv6, see &quot;Dual Stack IPv6 Configuration Task Flow&quot; chapter in the <a href="#">System Configuration Guide for Cisco Unified Communications Manager</a>.</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 <a href="#">Multilevel Precendence and Preemption Task Flow, on page 787</a>.</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>
<tr>
<td>Transport Layer Signalling (TLS)</td>
<td>Transport Layer Security (TLS) provides secure and reliable signaling and data transfer between two systems or devices, by using secure ports and certificate exchange. For more information to configure TLS, see the &quot;TLS Setup&quot; chapter in the <a href="#">Security Guide for Cisco Unified Communications Manager</a>.</td>
</tr>
<tr>
<td>V.150</td>
<td>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). For more information to configure V.150, see the &quot;Cisco V.150 Minimum Essential Requirements (MER)&quot; chapter in the <a href="#">Security Guide for Cisco Unified Communications Manager</a>.</td>
</tr>
</tbody>
</table>
CHAPTER 66

Configure Multilevel Precedence and Preemption

- Multilevel Precedence and Preemption Overview, on page 787
- Multilevel Precedence and Preemption Prerequisites, on page 787
- Multilevel Precedence and Preemption Task Flow, on page 787
- Multilevel Precedence and Preemption Interactions and Restrictions, on page 802

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 Precedence 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 789, 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 790</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure a Resource Priority Namespace Network Domain, on page 790</td>
<td></td>
</tr>
</tbody>
</table>
### 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 791</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 791</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure the Enterprise Parameters for Multilevel Precedence and Preemption, on page 792</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 MLPP-related enterprise parameters apply to these devices and common device configurations.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure a Partition for Multilevel Precedence and Preemption, on page 793</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 DN and route patterns. These entities associate with DN and users dial. For simplicity, partition names usually reflect their characteristics.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure a Calling Search Space for Multilevel Precedence and Preemption, on page 794</td>
<td>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 6</strong> Configure a Route Pattern for Multilevel Precedence and Preemption, on page 795</td>
<td>Configure route patterns to route or block both internal and external calls.</td>
</tr>
<tr>
<td><strong>Step 7</strong> Configure a Translation Pattern for Multilevel Precedence and Preemption, on page 796</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 8</strong> Configure Multilevel Precedence and Preemption for Gateways, on page 797</td>
<td>Configure Cisco Unified Communications Manager to communicate with non-IP telecommunications devices.</td>
</tr>
<tr>
<td><strong>Step 9</strong> Configure Multilevel Precedence and Preemption for Phones, on page 798</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>---------------------------------------------------------------------------------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Step 10 Configure a Directory Number to Place Multilevel Precedence and Preemption Calls, on page 800</td>
<td>After you configure a device, you can add a line (directory number) from the updated Device Configuration window.</td>
</tr>
<tr>
<td>Step 11 Configure a User Device Profile for Multilevel Precedence and Preemption, on page 800</td>
<td>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.</td>
</tr>
<tr>
<td>Step 12 Configure the Default Device Profile for Multilevel Precedence and Preemption, on page 801</td>
<td>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.</td>
</tr>
</tbody>
</table>

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>Step 1 Configure a Multilevel Precedence and Preemption Domain, on page 790</td>
<td>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.</td>
</tr>
<tr>
<td>Step 2 Configure a Resource Priority Namespace Network Domain, on page 790</td>
<td>Configure namespace domains for a Voice over Secured IP (VoSIP) network that uses SIP trunks. Your system prioritizes the SIP-signaled</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

Step 1  From Cisco Unified CM Administration, choose System > MLPP > Domain > MLPP Domain.

Step 2  Click Add New.

Step 3  In the Domain Name 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 (_).

Step 4  In the Domain ID field, enter a unique six-character hexadecimal MLPP domain ID.

Domain IDs must fall in the range between 000001 and FFFFFFF. (000000 is reserved for the default MLPP domain ID.)

Step 5  Click Save.

Configure a Resource Priority Namespace Network Domain

Configure namespace domains 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.
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>
<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>
<tr>
<td>Parameter</td>
<td>Description</td>
</tr>
<tr>
<td>-----------</td>
<td>-------------</td>
</tr>
<tr>
<td>Precedence Alternate Party Timeout</td>
<td>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.</td>
</tr>
<tr>
<td>Use Standard VM Handling For Precedence Calls</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. **Step 1** From Cisco Unified CM Administration, choose Call Routing > Class of Control > Partition.
2. **Step 2** Click Add New to create a new partition.
3. **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.
4. **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.
5. **Step 5** To create multiple partitions, use one line for each partition entry.
6. **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.
7. **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 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 77: 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 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>
### MLPP Precedence

Choose an MLPP precedence setting for this route pattern from the drop-down list box:
- 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.

### Resource Priority Namespace Network Domain


---

### 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**

1. **Step 1**
   - In Cisco Unified CM Administration, choose Call Routing > Translation Pattern.

2. **Step 2**
   - Perform one of the following tasks:
     - To modify the settings for an existing translation pattern, enter search criteria, click Find, and choose an existing Translation Pattern from the resulting list.
• To add a new translation pattern, click Add New.

**Step 3**
From the **MLPP Precedence** drop-down list, choose one of the following settings for this translation pattern:
- **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.

**Step 4**
From the **Resource-Priority Namespace Network Domain** drop-down list, choose a resource priority namespace network domain that you configured.

**Step 5**
From the **Calling Search Space** drop-down list, choose the calling search space that you configured.

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

---

**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.
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>
<thead>
<tr>
<th>MLPP Settings for Phones Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<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 <strong>None</strong> 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 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.

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.
• 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 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 6  Click Save.

## Multilevel Precedence and Preemption Interactions and Restrictions

### Multilevel Precedence and Preemption Interactions

#### Table 80: 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 Feature

MLPP interacts 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

Immediate Divert diverts calls to voice-messaging mail boxes 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.

Resource Reservation Protocol (RSVP)

RSVP supports MLPP inherently. The Cisco Unified Communications Manager System Guide explains how MLPP functions when RSVP is activated.

Supplementary Services

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 81: 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>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<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>
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 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 XV

SIP Interoperability

• Configure SIP Normalization and Transparency, on page 809
• Configure SDP Transparency Profiles, on page 815
Configure SIP Normalization and Transparency

• SIP Normalization and Transparency Overview, on page 809
• SIP Normalization and Transparency Prerequisites, on page 810
• SIP Normalization and Transparency Configuration Task Flow, on page 810

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-PAIpassthrough**—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.
- **cisco-meeting-server-interop**—Provides interoperability between Cisco Unified Communications Manager and Cisco Meeting Server (CMS).
- **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 810.


SIP Normalization and Transparency Configuration Task Flow

<table>
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<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Create New SIP Normalization and Transparency Scripts, on page 811</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</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
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<tr>
<td>-------------------------------------------------------</td>
<td>-------------------------------------------------------------------------</td>
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</tr>
<tr>
<td>script in the SIP Normalization Script Configuration window or you can import a customized script.</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> Apply Normalization or Transparency Script to SIP Trunk, on page 812</td>
<td>In the Trunk Configuration window, apply a script directly to a SIP trunk. Cisco Unified Communications Manager applies the script to all the SIP messaging that passes through the trunk.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> Apply Normalization or Transparency to SIP Devices, on page 812</td>
<td>If you want to apply a normalization or transparency script to a SIP line, apply a script to the SIP profile that is associated to that SIP line. Cisco Unified Communications Manager applies the script to all SIP messaging that uses that SIP profile.</td>
<td></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**

**Step 1** In Cisco Unified CM Administration, choose Device > Device Settings > SIP Normalization Script.

**Step 2** Click Add New.

The SIP Normalization Script Configuration window appears.

**Step 3** Enter a Name and Description for your script.

**Step 4** If you are writing a new script, edit the script in the Contents text box.

**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 box, 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 (+) button 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 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, choose a profile.

Step 4  From the **Normalization Script** drop-down list box, 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 (+) button 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 68

Configure SDP Transparency Profiles

- SDP Transparency Profile Overview, on page 815
- SDP Transparency Profile Restrictions, on page 815
- SDP Transparency Profile Prerequisites, on page 816
- Configure SDP Transparency Profile, on page 816

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

<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; Device Settings &gt; SDP Transparency Profile.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Enter a Name and Description.</td>
</tr>
<tr>
<td>Step 4</td>
<td>In the Attribute Information pane, create the rules for the SDP attributes that you want to pass through:</td>
</tr>
<tr>
<td></td>
<td>• 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 box, select Property.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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 box. 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.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click the (+) to create new lines where you can enter additional SDP attributes for this transparency profile.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

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.