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Feature Configuration Overview

• About this Guide, page 3
• Feature Quick Reference, page 3
• Generate a Phone Feature List, page 7

About this Guide

This guide provides information about the tasks that you need to complete in order to configure features on the Cisco 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 the Cisco Unified Communications Manager System Configuration Guide.

Feature Quick Reference

The following table provides an alphabetical list of the features described in this document, and lists the sections in this guide where you can find full configuration information about each of them. For information about how to determine which features are supported by your phones, see the Related Topics section below.

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<td>Receiving Calls</td>
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</table>
Generate a Phone Feature List

Generate a Phone Feature List report to determine which devices support the feature that you want to configure.

Procedure

Step 1  From Cisco Unified Reporting 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 already 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.
   The report is generated.
Configuration Tools

- Configuration Tools Overview, page 9
- Cisco Unified Communications Manager Administration, page 9
- Cisco Unified Communications Manager Serviceability, page 10

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 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 Cisco Unified Communications Manager Bulk Administration Tool (BAT) to make a large number of configuration changes at the same time. For more information, see Cisco Unified Communications Manager Bulk Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

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 Cisco Unified Communications Manager in the main window. For example, Cisco Unified Communications Manager may identify the following situations:
Cisco Unified Communications Manager currently operates with starter (demo) licenses, so upload the appropriate license files.

• Cisco Unified Communications Manager currently operates with an insufficient number of licenses, so upload additional license files.

• Cisco 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 Communications Manager Administration.

**Procedure**

**Step 1** Start your preferred operating system browser.

**Step 2** In the address bar of the web browser, enter the following case-sensitive URL:

https://<Unified CM-server-name>:{8443}/ccmadmin/showHome.do

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

**Note** You can optionally specify a port number.

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

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

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

---

**Cisco Unified Communications Manager Serviceability**

Some procedures in this guide require you to use the Cisco Unified Serviceability application to start or restart services on the Cisco 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 as a result of the user action. This functionality supports the Information Assurance feature of Cisco 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 Cisco Unified Communications Manager, IM and Presence Service 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.

• Cisco Unified Communications Manager only: Generates Cisco 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** At the main Cisco Unified Communications Manager Administration window, choose **Cisco Unified Serviceability** from the Navigation menu.

**Step 5** Enter the username and password that you specified during Cisco 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.
PART II

Remote Worker Features

- Cisco Unified Mobility, page 15
- Device Mobility, page 35
- Extend and Connect, page 43
- Remote Worker Emergency Calling, page 53
Cisco Unified Mobility Overview

Cisco Unified Mobility allows users to answer incoming calls on the desk phone or mobile phone, and to pick up in-progress calls on the desk phone or mobile phone without losing the connection. This set of features provides a single enterprise number that can be used on up to ten devices, as well as access to enterprise features through dual-tone multifrequency (DTMF) feature access codes.

Caution

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 Prerequisites

- Cisco Unified Mobile Voice Access service, which runs on only the publisher node.
- Cisco Unified Communications Manager Locale Installer (if you want to use non-English phone locales or country-specific tones).
## Cisco Unified Mobility Configuration Task Flow

### Procedure

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<td><strong>Step 2</strong></td>
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<td>Activate this service on the first node in the cluster.</td>
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<tr>
<td><strong>Step 3</strong></td>
<td>Configure a Softkey Template for Mobility, on page 17</td>
<td>Configure a Mobility Softkey to associate to mobility users so they can use mobility features from their deskphone.</td>
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<tr>
<td><strong>Step 4</strong></td>
<td>To Configure Single Number Reach, on page 18, perform the following subtasks:</td>
<td>Configure Single Number Reach for users to allow deskphones to extend calls to mobile devices.</td>
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<tr>
<td><strong>Step 5</strong></td>
<td>To Configure Cisco Jabber Clients for Mobility, on page 22, perform the following subtasks:</td>
<td>Configure Jabber for mobility to let users access enterprise communications features through a Jabber client on their smartphone.</td>
</tr>
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<tr>
<td><strong>Step 6</strong></td>
<td>To Configure Mobile Voice Access, on page 25, perform the following subtasks:</td>
<td>Optional. Configure Mobile Voice Access to provide interactive voice response (IVR) system to initiate two-stage dialed calls through the enterprise and activate or deactivate Cisco Unified Mobility capabilities.</td>
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### Activate the Cisco Unified Mobile Voice Access Service

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<td>Step 1</td>
<td>From Cisco Unified Serviceability, choose <strong>Tools &gt; Service Activation</strong>.</td>
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<td>Step 2</td>
<td>From the <strong>Server</strong> drop-down list, choose the publisher node.</td>
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<tr>
<td>Step 3</td>
<td>Click <strong>Go</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Under <strong>CM Services</strong>, check the <strong>Cisco Unified Mobile Voice Access Service</strong> check box.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
<td></td>
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</table>

### Configure a Softkey Template for Mobility

Configure a Mobility Softkey to associate to mobility users so they can use mobility features from their deskphone. The call states that you must configure are On Hook and Connected.

**Note**

Do not configure both the Mobility softkey and the MOVE softkey on the same template. For more information about which softkey your device uses, see your phone configuration guide.

**Before You Begin**

 Activate the Cisco Unified Mobile Voice Access Service, on page 17

**Procedure**

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<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose <strong>Device &gt; Device Settings &gt; Softkey Template</strong>.</td>
<td></td>
</tr>
</tbody>
</table>
The **Softkey Template Configuration** window appears.

**Step 2** Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.
d) Click **Save**.

**Step 3** Perform this step to add softkeys to an existing template.
a) Enter search criteria and click **Find**.
b) Choose an existing template.
The **Softkey Template Configuration** window appears.

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

**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. See the What to Do Next section for more information.

---

**What to Do Next**

- **Configure Single Number Reach**, on page 18
- **Configure Cisco Jabber Clients for Mobility**, on page 22

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**Configure Single Number Reach**

**Before You Begin**

- **Activate the Cisco Unified Mobile Voice Access Service**, on page 17
- **Configure a Softkey Template for Mobility**, on page 17
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<td>Configure users for mobility to specify which features they can use and the number of remote destinations they can use.</td>
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<tr>
<td>Step 2</td>
<td>Configure IP Phone for Mobility, on page 20</td>
<td>Configure an IP phone for mobility so that incoming calls can reach a user's remote destination.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure a Remote Destination Profile, on page 20</td>
<td>Configure remote destination profiles to apply settings to all of the numbers that you add as remote destinations for the user.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure a Remote Destination, on page 21</td>
<td>Configure remote destinations which are phones that are available for Cisco Unified Mobility call answer, call pickup, and can use Mobile Voice Access and Enterprise Feature Access for two-stage dialing.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure an Access List, on page 22</td>
<td>Optional. Configure an Access List if you want to allow or block certain calls to a user's remote destination.</td>
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### Configure a User for Mobility

Configure users for mobility to specify which features they can use and the number of remote destinations they can use.

#### Before You Begin

Configure a Softkey Template for Mobility, on page 17

### Procedure

**Step 1**

From Cisco Unified Communications Manager 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 choose an existing user from the resulting list.
To add a new user, click Add New.

Step 3  Check the Enable Mobility check box.
Step 4  Check the Enable Mobile Voice Access check box.
Step 5  In the Maximum Wait Time for Desk Pickup field, enter the maximum time in milliseconds that can pass before the user must pick up a call that is transferred from the mobile phone to the desk phone. Enter 0 to disable the desk phone transfer behavior.
Step 6  In the Remote Destination Limit field, enter the number of remote numbers that a user is permitted to have for Single Number Reach (SNR) targets.
Step 7  Click Save.

What to Do Next
Configure IP Phone for Mobility, on page 20

Configure IP Phone for Mobility

Before You Begin

• Configure a Softkey Template for Mobility, on page 17
• Configure a User for Mobility, on page 19

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Perform one of the following tasks:
        • To modify the settings for an existing phone, enter search criteria, click Find, and choose an existing phone from the resulting list.
        • To add a new phone, click Add New and choose the phone from the Phone Type drop-down list.
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  Select Save.

What to Do Next
Configure a Remote Destination Profile, on page 20

Configure a Remote Destination Profile

The settings that you enter in this profile will apply to all of the numbers that you add as remote destinations for the user.
Before You Begin
Configure IP Phone for Mobility, on page 20

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose Device > Device Profile > Remote Destination Profile.
Step 2 Click Add New.
Step 3 Configure the fields in the Remote Destination Profile Configuration window. See the online help for more information about the fields and their configuration options.
Step 4 Click Save.
Step 5 Under Association Information, click Add a New DN.
Step 6 In the Directory Number field, add the directory number of the desk phone that you want to mirror. The shared line is added and the Remote Destination Profile appears as an associated device.

What to Do Next
Configure a Remote Destination, on page 21

Configure a Remote Destination

Remote destinations are phones that are available for Cisco Unified Mobility call answer, call pickup, and that can use Mobile Voice Access and Enterprise Feature Access for two-stage dialing. Remote destinations can include any of the following devices:

• Mobile phones (single-mode and dual-mode)
• Enterprise numbers that are not on the same cluster as the desk phone
• Phone numbers on the PSTN

Before You Begin
Configure a Remote Destination Profile, on page 20

Procedure

Step 1 From Cisco Unified Communications Administration, choose Device > Remote Destination.
Step 2 Click Add New.
Step 3 Configure the fields on the Remote Destination Configuration window. See the online help for more information about the fields and their configuration options.
Step 4 Click Save.
Configure an Access List

Configure an Access List if you want to allow or block certain calls to a user’s remote destination.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Class of Control > Access List.
Step 2 In the Name field, enter a name.
Step 3 In the Description field, enter a description.
Step 4 Associate the access list to a user by choosing an ID from the Owner drop-down list.
Step 5 Click 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 Choose Save.
Step 7 Select Add Member.
Step 8 From the Filter Mask drop-down list, select 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 select this option, add a number or number string in the DN Mask field.

Step 9 Choose Save.
Step 10 Open the Remote Destination that you created.
Step 11 Perform one of the following steps:
   • If you created an Allowed access list, click the Ring this destination only if caller is in radio button and select 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 select the access list that you created from the drop-down list.

Configure Cisco Jabber Clients for Mobility

Before You Begin

• Activate the Cisco Unified Mobile Voice Access Service, on page 17
• Configure a Softkey Template for Mobility, on page 17
### Configure Mobility Profile

Assign mobility profiles to the mobility identity for Jabber client devices. While not required, the mobility profile indicates the caller ID that the system sends during setup of the Dial via Office Reverse (DVO-R) callback call leg to the mobility identity or alternate callback number. 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.

### Before You Begin

- Activate the Cisco Unified Mobile Voice Access Service, on page 17
- Configure a Softkey Template for Mobility, on page 17

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### Procedure

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<td><strong>Step 1</strong> Configure Mobility Profile, on page 23</td>
<td>Optional. Assign mobility profiles to the mobility identity for Jabber client devices to indicate the caller ID that the system sends during the Dial via Office Reverse (DVO-R) callback call leg to the mobility identity or alternate callback number. 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.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure a Dual-Mode Device, on page 24</td>
<td>Configure a dual-mode device to enable Dial via Office so that users can make calls from their mobile device as if they were dialing from their desk phone.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure Mobile Identity, on page 25</td>
<td>The mobile identity is the mobile phone number of the iPhone or Android device. If needed, you can also set the voicemail policy behavior.</td>
</tr>
</tbody>
</table>

---

### What to Do Next

Configure a Dual-Mode Device, on page 24
Configure a Dual-Mode Device

Configure a dual-mode device to enable Dial via Office so that users can make calls from their mobile device as if they were dialing from their desk phone.

**Before You Begin**

Configure Mobility Profile, on page 23

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>From Cisco Unified Communications Manager Administration, choose Device &gt; Phone.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click Add New.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>From the Phone Type drop-down list, choose Cisco Dual Mode for Android or Cisco Dual Mode for iPhone.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Click Next.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure the fields in the Phone Configuration window. See the Related Topics section for more information about the fields and their configuration options. For Product Specific Configuration Layout fields, see your Jabber client documentation.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

**What to Do Next**

Configure Mobile Identity, on page 25

**Related Topics**

Dual-Mode Device Configuration Fields, on page 24

**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 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>
<tr>
<td>Device Security Profile</td>
<td>Choose the security profile to apply to the device. You must apply a security profile to all phones that are configured in Cisco Unified Communications Manager Administration. To enable security features for a phone, you must configure a new security profile for the device type and protocol and apply it to the phone.</td>
</tr>
<tr>
<td>Rerouting Calling Search Space</td>
<td>Choose a calling search space for routing calls to configured remote destinations and mobility identities that are configured for this device.</td>
</tr>
</tbody>
</table>
Configure Mobile Identity

The mobile identity is the mobile phone number of the iPhone or Android device. If needed, you can also set the voicemail policy behavior.

**Before You Begin**

Configure a Dual-Mode Device, on page 24

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified Communications Manager Administration, choose <strong>Device &gt; Phone</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Find the dual-mode device that you created.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click <strong>Add New Mobile Identity</strong>.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure the fields on the <strong>Mobile Identity Configuration</strong> window. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
</tbody>
</table>

Configure Mobile Voice Access

When initiating a mobility call from a mobile device, users can use Mobile Voice Access to initiate calls from a mobile phone as if dialing from the desk phone.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 <strong>Configure Service Parameters for Mobile Voice Access</strong>, on page 25</td>
<td></td>
</tr>
<tr>
<td>Step 2 <strong>Configure Directory Number for Mobile Voice Access</strong>, on page 26</td>
<td></td>
</tr>
</tbody>
</table>

Configure Service Parameters for Mobile Voice Access

**Before You Begin**

Configure a User for Mobility, on page 19
Procedure

**Step 1** From Cisco Unified Communications 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 **Mobile Voice Access** service parameter to **True**.

**Step 5** For the **Mobile Voice Access Number** service parameter, enter the access number that users will dial to access this feature.

**Configure Directory Number for Mobile Voice Access**

Procedure

**Step 1** From Cisco Unified Communications Manager Administration, select **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 and 24 digits in length. Valid values are 0 to 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**.

**Configure Enterprise Feature Access**

Configure Enterprise Feature Access to enable access to hold, resume, transfer, and conference features from remote destinations.

Procedure

**Step 1** From Cisco Unified Communications 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 **Enable Enterprise Feature Access** service parameter to **True**.

**Step 5** (Optional) Modify the Enterprise Feature Access codes.

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

**Step 6**  From Cisco Unified Communications Manager Administration, choose **Call Routing > Mobility > Enterprise Feature Access**.

**Step 7**  In the **Number** field, enter a unique DID number that is required for enterprise feature access. This number supports transfer, conference, resume, and two-stage dialing from remote destinations.

**Step 8**  From the **Route Partition** drop-down list, choose the partition of the DID that is required for enterprise feature access.

**Step 9**  (Optional) In the **Description** field, enter a description of the Mobility Enterprise Feature Access number.

**Step 10**  (Optional) Check the **Default Enterprise Feature Access Number** check box to make this Enterprise Feature Access number the default for this system.

**Step 11**  Click **Save**.

---

**Configure Handoff Mobility**

Handoff Mobility allows you to configure a handoff number and partition for dual-mode phones between the Wi-Fi and mobile voice or cellular network.

**Procedure**

**Step 1**  From Cisco Unified Communications Manager Administration, choose **Call Routing > Mobility > Handoff Configuration**.

**Step 2**  In the **Handoff Number** field, enter the direct inward dialing (DID) number for handoff between the Wi-Fi and mobile voice or cellular network.

For numbers that start with the international escape character (+), you must precede the + with a backslash (\). Example: \+15551234.

**Step 3**  From the **Route Partition** drop-down list, choose the partition to which the handoff DID number belongs.

**Step 4**  Click **Save**.
Configure the Mobility Service Parameters

Procedure

Step 1  From Cisco Unified Communications Manager 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  Configure the service parameters that are listed under Clusterwide Parameters (System - Mobility). Click the name of the service parameter, which provides a hyperlink to a complete definition of the service parameter and its configuration options.

Cisco Unified Mobility Interactions and Restrictions

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 Auto Call Pickup Enabled service parameter is set to True, users must press only the PickUp softkey to pick up a call. If the service parameter is set to False, users must press the PickUp, GPickUp, or OPickUp softkey and then the Answer softkey.</td>
</tr>
</tbody>
</table>
| Automatic Alternate Routing | Cisco Unified Mobility supports Automatic Alternate Routing (AAR) as follows:  
• If a rejection occurs because of a lack of bandwidth for the location-based service, the rejection triggers AAR for any device that is configured for AAR.  
• If a rejection occurs because of Resource Reservation Protocol (RSVP), however, AAR is not triggered for calls to remote destinations. |
| Extend and Connect  | 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. For more information, see the “Extend and Connect” chapter. |
Interaction Feature

If External Call Control is configured, Cisco Unified Communications Manager follows the route decision from the adjunct route server for these Cisco Unified Mobility features:

- Cisco Unified Mobility
- Mobile Voice Access
- Enterprise Feature Access
- Dial via Office

Cisco Unified Communications Manager does not send a routing query for the following Cisco Unified Mobility features:

- Cell pickup
- Desk pickup
- Session handoff

Intelligent Session Control and Session Handoff

For direct calls to remote destinations that get anchored to the enterprise number, mobile users can use the Session Handoff feature to hand off the call to their deskphones.

You must enable Cisco Unified Mobility before you implement Intelligent Session Control.

Licensing

Cisco Unified Mobility is included in all user-based licenses from basic to professional.

Local Route Groups

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.

Number of Supported Calls

Each remote destination supports a maximum of two active calls. For Cisco Unified Mobility, each remote destination supports a maximum of two active calls through Cisco Unified Communications Manager. Using the Enterprise Feature Access directory number (DID number) to transfer or conference with DTMF is one call. When a Cisco Unified Mobility user receives a call while the user has two active 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 does not reach the remote destination. Instead, the call goes to the enterprise voicemail, if Call Forward No Answer (CFNA) is configured or if the call is not answered on a shared line.

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).
## Restrictions

<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</td>
<td>You do not need to configure settings for Call Forward Unregistered if you configured remote destinations for users. Call forwarding is handled as part of the Cisco Unified Mobility process.</td>
</tr>
<tr>
<td>Call Queuing</td>
<td>Cisco Unified Communications Manager does not support Call Queuing with Cisco Unified Mobility.</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>DND 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:</td>
</tr>
</tbody>
</table>
### Dual-Mode Phones

**Dual-Mode Handoff and Caller ID**

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.

**Dual-Mode Phones and CTI Applications**

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.

**Dual-Mode Phones and SIP Registration Period**

For dual-mode phones, Cisco 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 Device determines the registration period as defined by the Time Register Expires field in that profile.

### Enterprise Features From Cellular Networks

Enterprise features from cellular networks require out-of-band DTMF.

When using intercluster DNs 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.

### 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."
### Restriction | Description
--- | ---
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.

Multiple-Node Cluster Environment | 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.

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

QSIG | Mobility does not support QSIG (Q Signaling).

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.

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.
<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP URI and Direct Calls to Remote Destination</td>
<td>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.</td>
</tr>
<tr>
<td>Video Calls</td>
<td>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.</td>
</tr>
</tbody>
</table>

**Related Topics**

Ad Hoc Conferencing Service Parameters, on page 189

---

**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.

**Solution**

- 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`. 
Device Mobility Overview

The Device Mobility feature allows devices to assume settings based on their location. Cisco Unified Communications Manager uses the device IP subnets to determine the exact location of the device. By enabling Device Mobility within a cluster, you allow mobile users roam from one site to another and acquire the site-specific settings. Cisco Unified Communications Manager then uses these dynamically allocated settings for functions such as call routing, codec selection, and media resource selection.

Device Mobility Prerequisites

- Cisco Database Layer Monitor service running on the same node as the Cisco CallManager service
- Cisco TFTP service 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

### 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 7</td>
<td>Generate a report to identify devices that support the Device Mobility feature.</td>
</tr>
</tbody>
</table>
| Step 2 | To Enable Device Mobility, on page 36, perform one or both of the following subtasks:  
• Enable Device Mobility, on page 37  
• Configure Device Mobility for a Device, on page 37 | Perform these procedures as needed for your configuration. You can configure Device Mobility to be enabled across your cluster, for individual devices, or a combination of enabled and disabled for devices. |
| Step 3 | Configure Physical Location, on page 38 | Cisco Unified Communications Manager uses the geographic location to determine which network resources to assign to a device. If a user moves away from the home location, the system ensures that the device uses local media resources and the correct bandwidth for the call. |
| Step 4 | Configure Device Mobility Group, on page 38 | The device mobility group defines a logical group of sites with similar dialing patterns (for example, US_dmg and EUR_dmg). Depending on the network size and scope, your device mobility groups can represent countries, regions, states, provinces, or cities. |
| Step 5 | Configure Device Mobility Information, on page 39 | Device Mobility Information refers to the subnets and device pools that are used for Device Mobility. When a phone registers with Cisco Unified Communications Manager, the system compares the IP address of the device to the subnets that are configured for Device Mobility in the Device Mobility Info Configuration window. |
| Step 6 | Configure a Device Pool for Device Mobility, on page 40 | Unified Communications Manager assigns a device pool to an IP phone based on the device's IP subnet. |
| Step 7 | View Roaming Device Pool Parameters, on page 41 | Optional. Follow this procedure if you want to view and verify the current Device Mobility settings for a device. |

### Enable Device Mobility

Perform these procedures as needed for your configuration. You can configure Device Mobility to be enabled across your cluster, for individual devices, or a combination of enabled and disabled for devices.
## Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Enable Device Mobility, on page 37</td>
<td>Enable the Device Mobility service parameter if you want to enable Device Mobility on all devices in the cluster.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Device Mobility for a Device, on page 37</td>
<td>Follow this procedure if you want to enable Device Mobility for an individual device or if you want to specify a different Device Mobility value for individual devices.</td>
</tr>
</tbody>
</table>

### Enable Device Mobility

Enable the Device Mobility service parameter to enable Device Mobility on all devices in the cluster.

#### Procedure

**Step 1** From Cisco Unified CM Administration, click **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** To enable the **Device Mobility Mode** service parameter for all devices in the cluster, choose **On**.
**Step 5** Click **Save**.

### What to Do Next

- Configure Device Mobility for a Device, on page 37
- Configure Physical Location, on page 38

### Configure Device Mobility for a Device

When you enable Device Mobility mode in the Phone Configuration window, the Device Mobility Mode phone settings take priority over the Device Mobility Mode service parameter setting.

#### Procedure

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone**.
**Step 2** Perform one of the following tasks:

- To modify the settings for an existing device, enter search criteria, click **Find**, and choose an existing device from the resulting list.
• To add a new device, click **Add New**, and choose a device from the **Phone Type** drop-down list.

**Step 3**  
From the **Device Mobility Mode** drop-down list, choose a value.

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

---

**What to Do Next**

**Configure Physical Location, on page 38**

---

**Configure Physical Location**

Cisco Unified Communications Manager uses the geographic location to determine which network resources to assign to a device. If a user moves away from the home location, the system ensures that the device uses local media resources and the correct bandwidth for the call.

**Before You Begin**

**Enable Device Mobility, on page 36**

**Procedure**

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

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

**Step 3**  
In the **Name** field, enter a name to identify the physical location.  
The name can contain up to 50 alphanumeric characters with any combination of spaces, periods (.), hyphens (-), and underscores (_).

**Step 4**  
In the **Description** field, enter a description of the physical location.  
The description can include up to 50 characters in any language, but it cannot include double quotation marks ("), percentage sign (%), ampersand (&), or angle brackets (< >).

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

---

**What to Do Next**

**Configure Device Mobility Group, on page 38**

---

**Configure Device Mobility Group**

The device mobility group defines a logical group of sites with similar dialing patterns (for example, US_dmg and EUR_dmg). Depending on the network size and scope, your device mobility groups can represent countries, regions, states, provinces, or cities. For example, an enterprise with a worldwide network can choose device mobility groups that represent individual countries, whereas an enterprise with a national or regional network can define device mobility groups that represent states, provinces, or cities.

**Before You Begin**

**Configure Physical Location, on page 38**
Procedure

Step 1  From Cisco Unified CM Administration, choose System > Device Mobility > Device Mobility Group.
Step 2  Click Add New.
Step 3  In the Name field, enter a name to identify the device mobility group.
Step 4  In the Description field, enter the description of the profile.
          The description can include up to 50 characters in any language, but it cannot include double quotation marks ("), percentage signs (%), ampersands (&), or angle brackets (<>).
Step 5  Click Save.

What to Do Next
Configure Device Mobility Information, on page 39

Configure Device Mobility Information

Device Mobility Information refers to the subnets and device pools that are used for Device Mobility. When a phone registers with Cisco Unified Communications Manager, the system 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.

If the device pool in the phone record matches the device pool in the matching subnet, the system detects the phone as in its home location, and the phone retains the parameters of its home device pool.

If the device pool in the phone record does not match the device pools in the matching subnet, the system detects the phone as roaming.

Before You Begin
Configure Device Mobility Group, on page 38

Procedure

Step 1  From Cisco Unified CM Administration, choose System > Device Mobility > Device Mobility Info.
Step 2  Click Add New.
Step 3  Configure the fields. See the online help for more information about the fields and their configuration options.
Step 4  Click Save.

What to Do Next
Configure a Device Pool for Device Mobility, on page 40
Configure a Device Pool for Device Mobility

Unified Communications Manager assigns a device pool to an IP phone based on the device's IP subnet. The following steps describe the behavior:

- The IP phone tries to register to Unified Communications Manager by sending its IP address in the registration message.
- Unified Communications Manager derives the device's IP subnet and matches it with the subnet configured in the Device Mobility Information.
- If the subnet matches, Unified Communications Manager provides the device with a new configuration based on the device pool configuration.

Before You Begin

Configure Device Mobility Information, on page 39

Procedure

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

Step 2 Perform one of the following steps:

- Click **Add New**.
- Click **Find** and choose an existing Device Pool.

Step 3 Configure the fields in the **Device Pool Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

Step 4 Click **Save**.

What to Do Next

View Roaming Device Pool Parameters, on page 41

Related Topics

Device Pool Fields for Device Mobility, on page 40

Device Pool Fields for Device Mobility

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Physical Location</td>
<td>From the drop-down list, choose the physical location for this device pool. The system uses physical location with the Device Mobility feature to identify the parameters that relate to a specific geographical location.</td>
</tr>
<tr>
<td>Device Mobility Group</td>
<td>From the drop-down list, choose the device mobility group.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Device Mobility Calling Search Space</td>
<td>From the drop-down list, choose the calling search space to be used as the device calling search space when the device is roaming and in same device mobility group.</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>From the drop-down list, choose the calling search space for the device to use when automated alternate routing (AAR) is performed. The AAR calling search space is the collection of route partitions that are searched to determine how to route a collected (originating) number that is otherwise blocked due to insufficient bandwidth.</td>
</tr>
<tr>
<td>AAR Group</td>
<td>From the drop-down list, choose the AAR group for this device. The AAR group provides the prefix digits that are used to route calls that are otherwise blocked due to insufficient bandwidth. An AAR group setting of <strong>None</strong> specifies that no rerouting of blocked calls is attempted.</td>
</tr>
</tbody>
</table>
| Calling Party Transformation CSS          | This setting allows you to localize the calling party number on the device. Ensure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device pool.  
**Note** Before the call occurs, the device must apply the transformation by using digit analysis. If you choose **None** and you check the Use Device Pool Calling Party Transformation CSS check box in the device configuration window, the transformation does not match and is not applied. Ensure that you configure the calling party transformation pattern in a partition that is not used for routing. |
| Called Party Transformation CSS           | This setting allows you to localize the called party number on the device. Make sure that the Called Party Transformation CSS that you choose contains the called party transformation pattern that you want to assign to this device pool.  
**Note** If you choose **None**, the transformation does not match and is not applied. Ensure that you configure the called party transformation pattern in a partition that is not used for routing.                                                                                           |

**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 and Restrictions

Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Normalization</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>
</tbody>
</table>
| 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 (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. |

Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| IP Address  | The Device Mobility feature depends on the IPv4 address of the device that registers with Cisco Unified Communications Manager.  
  - The phone must have a dynamic IPv4 address to use 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. |
| IPv6        | Device mobility supports only IPv4 addresses, so you cannot use phones with an IP addressing mode of IPv6 Only with device mobility. |
Extend and Connect Overview

The Extend and Connect feature allows administrators to deploy Unified Communications (UC) Computer Telephony Integration (CTI) applications that interoperable 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
- 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
- Persistent Connection
- Play Whisper Announcement
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

This section describes the procedures that you must complete to provision Cisco Unified Communications Manager users with Extend and Connect capabilities. For information about provisioning Cisco Jabber for Windows users with Extend and Connect, see the Cisco Jabber for Windows Environment Configuration Guide.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure User Account, on page 44</td>
<td>Enable mobility for users so that they can use CTI remote devices. CTI devices are off-cluster phones that work with Cisco UC applications.</td>
</tr>
<tr>
<td>Step 2 Add User Permissions, on page 45</td>
<td>Add access control group permissions.</td>
</tr>
<tr>
<td>Step 3 Create CTI Remote Devices, on page 46</td>
<td>Configure off-cluster phones that users can use with Cisco UC applications.</td>
</tr>
<tr>
<td>Step 4 Add Directory Number to a Device, on page 47</td>
<td>Associate a directory number with the CTI remote device.</td>
</tr>
<tr>
<td>Step 5 (Optional) Add Remote Destination, on page 47</td>
<td>Add a numerical address or directory URI that represents the other phones that the user owns.</td>
</tr>
<tr>
<td>Step 6 (Optional) Verify Remote Destination, on page 48</td>
<td>Verify if the remote destination is successfully added for a user.</td>
</tr>
<tr>
<td>Step 7 Associate User with Device, on page 49</td>
<td>Associate an end user account to the CTI remote device.</td>
</tr>
<tr>
<td>Step 8 Create CCMIP Profile, on page 49</td>
<td>Retrieve device names and settings from Unified Communications Manager.</td>
</tr>
</tbody>
</table>

Configure User Account

For a new or existing users 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**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>User Management &gt; End User</strong>. The <strong>Find and List Users</strong> window appears.</th>
</tr>
</thead>
</table>
| Step 2 | Perform one of the following:  
  • To configure a new user, click **Add New**.  
  • To select an existing user, specify the appropriate filters in the **Find User Where** field, select **Find** to retrieve a list of users, and then select the user from the list. |
| **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**. |

What to Do Next

**Add User Permissions**, on page 45

**Add User Permissions**

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

**Before You Begin**

**Configure User Account**, on page 44

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>User Management &gt; End User</strong>. The <strong>Find and List Users</strong> window appears.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Specify the appropriate filters in the <strong>Find User Where</strong> field, and then select <strong>Find</strong> to retrieve a list of users.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Select the user from the list. The <strong>End User Configuration</strong> window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Locate the <strong>Permissions Information</strong> section.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Add to Access Control Group</strong>. The <strong>Find and List Access Control Groups</strong> window appears.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click <strong>Find</strong>. The Access Control Group list for Standard Users appears.</td>
</tr>
</tbody>
</table>
| Step 7 | Check the check boxes next to the following permissions:  
  • Standard CCM End-Users |
• Standard CTI Enabled

Step 8  Click Add Selected.
The window closes and the access control groups are added to the user account.

Step 9  Click Save.

What to Do Next
Create CTI Remote Devices, on page 46

Create CTI Remote Devices
Add User Permissions, on page 45
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.
The Find and List Phone window appears.

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. See the online help for more information about the fields and their configuration options.

Step 9  Click Save.
The fields to associate directory numbers and add remote destinations are displayed in the **Phone Configuration** window.

---

**What to Do Next**

Perform the following tasks:

- Add Remote Destination, on page 47
- Add Directory Number to a Device, on page 47

---

**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).

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

**Procedure**

1. **Step 1**
   - Locate the **Association Information** section in the **Phone Configuration** window.

2. **Step 2**
   - Click **Add a new DN**.
   - The **Directory Number Configuration** window appears.

3. **Step 3**
   - Specify a directory number in the **Directory Number** field.

4. **Step 4**
   - Configure all other required fields. See the online help for more information about the fields and their configuration options.

5. **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 users can add remote destinations through the Cisco Jabber interface. For more information, see the *Cisco Jabber for Windows Environment Configuration Guide*.

- 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 Communications Manager Administration, choose **Device > Phone**.
The **Find and List Phones** window appears.

**Step 2** Specify the appropriate filters in the **Find Phone Where** field and then select **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. See the online help for more information about the fields and their configuration options.

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

**Verify Remote Destination**

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

**Procedure**

**Step 1** From Cisco Unified Communications Manager Administration, choose **Device > Phone**.
The **Find and List Phones** window appears.

**Step 2** Specify the appropriate filters in the **Find Phone Where** field 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**.
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 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.

What to Do Next

Create CCMIP Profile, on page 49

Create CCMIP Profile

Cisco Jabber requires a Cisco IP Phone (CCMCIP) profile to retrieve device names and settings from Cisco Unified Communications Manager. For more information about CCMCIP profiles, see related topics in the Deployment Guide for IM and Presence Service on Cisco Unified Communications Manager.

Procedure

Step 1 Open the Cisco Unified CM IM and Presence Administration or Cisco Unified Presence Administration interface.
Step 2 Select Application > Cisco Jabber > CCMCIP Profile.
In some versions of Cisco Unified Presence, this path is as follows: Application > Cisco Unified Personal Communicator > CCMCIP Profile.

Step 3 Click Add New.

Step 4 Specify a name for the profile in the Name field.

Step 5 Specify the hostname or IP address of your primary Unified Communications Manager instance in the Primary CCMCIP Host field.

Step 6 Specify the hostname or IP address of your backup Unified Communications Manager instance in the Backup CCMCIP Host field.

Step 7 Leave the default value for Server Certificate Verification.

Step 8 Click Add Users to Profile.

Step 9 Add the appropriate users to the CCMCIP gateway profile.

Step 10 Click Add Selected.

Step 11 Click Save.

---

Extend and Connect Interactions and Restrictions

### 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. |
Interaction Feature

- The incoming caller ID information (name and number) is displayed on the Jabber client.
- This information may also be displayed on the device, depending on your carrier and trunk configuration.
- Outbound Dial Via Office calls to the remote destination display Voice Connect as the name and the trunk DID as the number.
- Configure the trunk DID in the Unified CM Trunk Pattern, Route Pattern, or Cisco Gateway. This configuration may also be assigned by the carrier. The number field may display as blank if the trunk DID is not configured.
- Outbound calls to the desired party display the CTI Remote Device Display Name and Directory Number (DN) as configured in Unified Communications Manager.
- Remote destination numbers are never displayed to the called party.

### Extend and Connect Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| Maximum number of remote destinations | You can configure up to ten remote destinations for each CTI remote device.  
**Note**  
By default, four remote destinations are supported per device. You can set the maximum number to 10 remote destinations per device.                                                                                           |
| Off-cluster devices                | - Remote destination numbers must represent off-cluster devices.  
- Remote destinations can be off-cluster URIs.                                                                                                                                                                                                                                   |
| Directory numbers                  | You cannot configure directory numbers as remote destination numbers.                                                                                                                                                                                                          |
| Cisco Jabber                       | Before you save the remote destinations that are configured using Cisco Jabber, verify if the remote destinations can be routed by the configured dial plan.                                                                                                                                 |
| Application dial rules             | Application Dial Rules are applied to all remote destinations that are configured on the CTI remote device through the Cisco Unified Communications Manager Administration interface and Cisco Jabber.  
**Note**  
Advise end users which number formats the Application Dial Rules are configured to support (for example, nn-nnn-nnnn, E.164, both).                                                                                              |
| Remote destination number          | Each remote destination number must be unique within the cluster.  
**Note**  
The same remote destination number cannot be used by two or more users.                                                                                                                                                                                                       |
<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>
</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 the Cisco Emergency Responder Administration Guide.
## Remote Worker Emergency Calling 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 User As a Remote Worker, on page 54</td>
<td>Associate the off-premises device with the owner of the device.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Specify Alternate Routing for Emergency Calling, on page 55</td>
<td>These parameters specify the calling search space and destination number that are used to restrict the routing of any call made from a registered off-premises device where the user chose not to set a location. If these parameters are not configured, calls are routed normally.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the Application Server, on page 55</td>
<td>Direct end users to the application server where they enter the location of the device.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure E911 Messages, on page 56</td>
<td>Configure the E911 messages that appear on an off-premises end-user phone.</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 information about configuring Intrado on the Cisco Emergency Responder, see the *Cisco Emergency Responder Administration Guide*.

#### Procedure

1. **Step 1**
   - From Cisco Unified Communications Manager Administration, choose Device > Phone. The Find and List Phones window is displayed.
2. **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.
3. **Step 3**
   - Choose the phone for which you want to configure Remote Worker Emergency Calling. The Phone Configuration window is displayed.
4. **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.
5. **Step 5**
   - Click Save.

#### What to Do Next

Specify Alternate Routing for Emergency Calling, on page 55
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.

**Before You Begin**

Configure User As a Remote Worker, on page 54

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified Communications Manager Administration, 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 is displayed.</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>

**What to Do Next**

Configure the Application Server, on page 55

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.

**Before You Begin**

Specify Alternate Routing for Emergency Calling, on page 55

**Procedure**

| Step 1 | In Cisco Unified CM Administration, choose System > Application Server. The Find and List Application Servers window is displayed. |
| Step 2 | Click Add New. |
The Application Server window is displayed.

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

What to Do Next
Configure E911 Messages, on page 56

Configure E911 Messages

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

Procedure

Step 1 In Cisco Unified Communications Manager 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 Optionally, you can edit the E911 messages to be displayed on off-premises devices.
Step 4 Click Save.
Remote Network Access

- Wireless LAN, page 59
- Wi-Fi Hotspot, page 63
- VPN Client, page 65
CHAPTER 7

Wireless LAN

- Wireless LAN Overview, page 59
- Wireless LAN Configuration Task Flow, page 59

Wireless LAN Overview

This feature removes the need for users to configure Wi-Fi parameters on their phones by allowing the administrator to configure Wi-Fi profiles for them. Devices can automatically download and apply the Wi-Fi configuration from Cisco Unified Communications Manager. You can configure a Network Access Profile, which contains information about VPN connectivity and HTTP proxy settings.

Wireless LAN 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 7</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 60</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 60</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure a Wireless LAN Profile Group, on page 61</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>To Link a Wireless LAN Profile Group to a Device or Device Pool, on page 61</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Link a Wireless LAN Profile Group to a Device, on page 61</td>
<td></td>
</tr>
</tbody>
</table>
Configure a Network Access 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. See the online help for more information about the fields and their configuration options.
Step 4 Click Save.

What to Do Next
Configure a Wireless LAN Profile, on page 60

Configure a Wireless LAN Profile

Before You Begin
(Optional) Configure a Network Access Profile, on page 60

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. See the online help for more information about the fields and their configuration options.
Step 4 Click Save.

What to Do Next
Configure a Wireless LAN Profile Group, on page 61

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Link a Wireless LAN Profile Group to a Device Pool, on page 62</td>
<td></td>
</tr>
</tbody>
</table>
Configure a Wireless LAN Profile Group

Before You Begin
Configure a Wireless LAN Profile, on page 60

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; Wireless LAN Profile Group.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the fields in the Wireless LAN Profile Group 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
Link a Wireless LAN Profile Group to a Device or Device Pool, on page 61

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

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Link a Wireless LAN Profile Group to a Device, on page 61</td>
</tr>
<tr>
<td>Step 2</td>
<td>Link a Wireless LAN Profile Group to a Device Pool, on page 62</td>
</tr>
</tbody>
</table>

Link a Wireless LAN Profile Group to a Device

Before You Begin
Configure a Wireless LAN Profile Group, on page 61

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or 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>Perform one of the following tasks:</td>
</tr>
<tr>
<td></td>
<td>• To modify the settings for an existing device, enter search criteria, click Find, and choose an existing device from the resulting list.</td>
</tr>
</tbody>
</table>
• To add a new device, 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.

**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, Cisco Unified Communications Manager uses the device pool setting.

**Before You Begin**

Configure a Wireless LAN Profile Group, on page 61

**Procedure**

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

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

• To modify the settings for an existing device pool, enter search criteria, click Find, and choose an existing device pool from the resulting list.

• To add a new device pool, click Add New.

**Step 3** From the Wireless LAN Profile Group drop-down list, choose a Wireless LAN Profile Group.

**Step 4** Click Save.
Wi-Fi Hotspot

• Wi-Fi Hotspot Overview, page 63
• Configure Wi-Fi Hotspot Profile, page 63

Wi-Fi Hotspot Overview

This feature allows users to use their desk phones to provide a Wi-Fi Hotspot, so that they can connect a Wi-Fi device such as a tablet or a smartphone to the network through the desk phone. The desk phones can automatically download the Wi-Fi Hotspot configuration from Cisco 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 Communications Manager Administration, select 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 <strong>Wi-Fi Hotspot Profile 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 Save.</td>
</tr>
</tbody>
</table>
CHAPTER 9

VPN Client

- VPN Client Overview, page 65
- VPN Client Prerequisites, page 65
- VPN Client Configuration Task Flow, page 65

VPN Client Overview

The Cisco VPN Client for Cisco Unified IP Phones creates a secure VPN connection for employees who telecommute. All settings of the Cisco VPN Client are configured through Cisco Unified CM 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 Cisco 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

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Complete Cisco IOS Prerequisites, on page 66</td>
<td>Complete Cisco IOS prerequisites. Perform this action if you want to configure Cisco IOS VPN.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Cisco IOS SSL VPN to Support IP Phones, on page 67</td>
</tr>
<tr>
<td>---------</td>
<td>---------------------------------------------------------------</td>
</tr>
<tr>
<td>Step 3</td>
<td>Complete ASA Prerequisites for AnyConnect, on page 69</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure the VPN concentrators for each VPN Gateway.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Upload VPN Concentrator Certificates, on page 69</td>
</tr>
<tr>
<td>Step 7</td>
<td>Configure VPN Gateway, on page 70</td>
</tr>
<tr>
<td>Step 8</td>
<td>Configure VPN Group, on page 72</td>
</tr>
<tr>
<td>Step 9</td>
<td>Perform one of the following:</td>
</tr>
<tr>
<td></td>
<td>• Configure VPN Profile</td>
</tr>
<tr>
<td></td>
<td>• Configure VPN Feature Settings</td>
</tr>
<tr>
<td>Step 10</td>
<td>Add VPN Details to Common Phone Profile, on page 76</td>
</tr>
<tr>
<td>Step 11</td>
<td>Upgrade the firmware for Cisco Unified IP Phones to a version that supports VPN.</td>
</tr>
<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**

Before you create Cisco IOS configuration for VPN client on an IP Phone, complete the following steps:
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.

What to Do Next
Configure Cisco IOS SSL VPN to Support IP Phones, on page 67

Configure Cisco IOS SSL VPN to Support IP Phones

Before You Begin
Complete Cisco IOS Prerequisites, on page 66

Procedure

Step 1 Configure Cisco IOS locally.
   a) Configure the Network Interface.
      Example:
      ```
      router(config)# interface GigabitEthernet0/0
      router(config-if)# description "outside interface"
      router(config-if)# ip address 10.1.1.1 255.255.255.0
      router(config-if)# duplex auto
      router(config-if)# speed auto
      router(config-if)# no shutdown
      router#show ip interface brief (shows interfaces summary)
      ```
   b) Configure static and default routes by using this command:
      `router(config)# ip route <dest_ip> <mask> <gateway_ip>`
      Example:
      ```
      router(config)# ip route 10.10.10.0 255.255.255.0 192.168.1.1
      ```

Step 2 Generate and register the CAPF certificate to authenticate the IP phones with an LSC.

Step 3 Import the CAPF certificate from Cisco Unified Communications Manager:
   a) From the Cisco Unified OS Administration, choose Security > Certificate Management.
      Note This location may change 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.
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 Cisco 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 >
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 Cisco Unified Communications Manager.

Example:

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

- Register the generated certificate with Cisco 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 Cisco 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.

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:

```
username CP-7975G-SEP001AE2BC16CB password k1kLQfQoxyC04ti9 encrypted
```

What to Do Next
Configure VPN concentrators for each VPN gateway.

Complete ASA Prerequisites for AnyConnect
Before you create an ASA configuration for VPN client on an IP phone, complete the following steps:

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 by executing the following command:
      ```
      show activation-key detail
      ```
   b) If necessary, obtain a new license with additional SSL VPN sessions and Linksys phone enabled.
Step 4 Ensure 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 vpnpool
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 Cisco Unified Communications Manager.
- If it 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, then disable host id check on Cisco Unified Communications Manager.

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 Cisco Unified Communications Manager using the...
procedure in this section. Cisco 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 Phones are trusted.

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

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>From Cisco Unified OS Administration, choose <strong>Security &gt; Certificate Management</strong>. The <em>Certificate List</em> window appears.</td>
</tr>
<tr>
<td>2</td>
<td>Click <strong>Upload Certificate</strong>. The <em>Upload Certificate</em> dialog box appears.</td>
</tr>
<tr>
<td>3</td>
<td>From the <strong>Certificate Purpose</strong> drop-down list, choose <strong>Phone-VPN-trust</strong>.</td>
</tr>
<tr>
<td>4</td>
<td>Click <strong>Browse</strong> to choose the file that you want to upload.</td>
</tr>
<tr>
<td>5</td>
<td>Click <strong>Upload File</strong>.</td>
</tr>
<tr>
<td>6</td>
<td>Choose another file to upload or click <strong>Close</strong>. For more information about certificate management, see Chapter 6, “Security,” in the <em>Cisco Unified Communications Operating System Administration Guide</em>.</td>
</tr>
</tbody>
</table>

**What to Do Next**

Configure VPN Gateway, on page 70

**Configure VPN Gateway**

To add, update, or copy a VPN gateway, perform the following procedure:

**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 69.

**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>Advanced Features &gt; VPN &gt; VPN Gateway</strong>.</td>
</tr>
</tbody>
</table>
| 2    | Perform one of the following tasks:  
  a) To add a new profile, click **Add New**.  
  b) To copy an existing VPN gateway, locate the appropriate profile, click the **Copy** button next to the VPN gateway that you want to copy. |
c) To update an existing profile, locate the appropriate VPN gateway and modify the settings. When you click Add New, the configuration window appears with the default settings for each field. When you click Copy, the configuration window appears with the copied settings.

**Step 3** Configure the fields in the **VPN Gateway Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

**Step 4** Click Save.

---

**What to Do Next**

Configure VPN Group, on page 72

**Related Topics**

VPN Gateway Fields for VPN Client, on page 71

---

### 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. **Note** 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 |
| 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 will fail to connect to that concentrator. **Note** 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

To add, update, or copy a VPN group, perform the following procedure:

**Before You Begin**
Configure VPN Gateway, on page 70

**Procedure**

**Step 1** In Cisco Unified Communications Manager Administration, choose *Advanced Features > VPN > VPN Group.*

**Step 2** Perform one of the following tasks:
   a) To add a new profile, click *Add New.*
   b) To copy an existing VPN group, locate the appropriate profile, click the *Copy* button next to the VPN group that you want to copy.
   c) To update an existing profile, locate the appropriate VPN group and modify the settings.
      When you click *Add New,* the configuration window appears with the default settings for each field. When you click *Copy,* the configuration window appears with the copied settings.

**Step 3** Configure the fields in the *VPN Group Configuration* window. See the Related Topics section for more information about the fields and their configuration options.

**Step 4** Click *Save.*

**What to Do Next**
Perform one of the following tasks:
   - Configure VPN Profile, on page 73
   - Configure VPN Feature Parameters, on page 74

**Related Topics**
VPN Group Fields for VPN Client, on page 72

**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>
</tbody>
</table>
Usetheupanddownarrowbuttonstomoveavailable
VPNgatewaysintoandoutofthisVPNgroup.
IftheVPNclientencountersacriticalerrorand
cannotconnecttoaparticularVPNgateway,itiswill
attempttomovetothenextVPNgatewayinthelist.

YoucanadduptoamaximumofthreeVPNgateways
toaVPNgroup.Also,thetotal
numberofcertificatesintheVPNgroup
cannotexceed10.

Note

Configure VPN Profile
Toadd,update,orcopyaVPNprofile,performthefollowingprocedure:

Procedure

**Step 1** InCiscoUnifiedCommunicationsManagerAdministration,choose*Advanced Features > VPN > VPN Profile*.

**Step 2** Performoneofthefollowingtasks:
a) Toaddnewprofile,clickAdd New.
b) Tocopyanexistingprofile,locatetheappropriateprofileandclicktheCopybuttonnexttotheVPN
profilethatyouwanttocopy.
c) Toupdateanexistingprofile,specifytheappropriatefiltersintheFind VPN Profile Where,clickFind,
andmodifythesettings.
WhenyouclickAdd New,theconfigurationwindowappearswiththedefaultsettingsforeachfield.WhenyouclickCopy,theconfigurationwindowappearswiththecopiedsettings.

**Step 3** ConfigurethefieldsintheVPN Profile Configurationwindow.SeetheRelatedTopicssectionformore
informationaboutthefieldsandtheirconfigurationoptions.

**Step 4** ClickSave.

What to Do Next

Add VPN Details to Common Phone Profile, on page 76

Related Topics

VPN Profile Fields for VPN Client, on page 73

VPN Profile Fields for VPN Client

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

<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>
</tbody>
</table>
### Configure VPN Feature Parameters

**Procedure**

**Step 1** In Cisco Unified Communications Manager Administration, choose Advanced Features > VPN > VPN Feature Configuration.

**Step 2** Configure the fields in the VPN Feature Configuration window. See the Related Topics section for more information about the fields and their configuration options.

**Step 3** Click Save.

**What to Do Next**

Perform the following tasks:

---

<table>
<thead>
<tr>
<th>Field</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<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 login attempt occurs, a user manually clears the password, or the phone resets or loses power.</td>
</tr>
</tbody>
</table>
• Upgrade the firmware for Cisco Unified IP Phones to a version that supports VPN. For more information about upgrading the firmware, see the *Cisco Unified IP Phone Administration Guide* for your Cisco Unified IP Phone model.

• Using a supported Cisco Unified IP Phone, establish the VPN connection.

**Related Topics**

[VPN Feature Parameters, on page 75](#)

### VPN Feature Parameters

The following table provides descriptions of the VPN feature configuration parameters.

*Table 1: VPN Feature Configuration 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:</td>
</tr>
<tr>
<td></td>
<td>Default: 1290 bytes</td>
</tr>
<tr>
<td></td>
<td>Minimum: 256 bytes</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> If it is non zero and less than the value specified in Cisco Unified Communications Manager, the keep alive setting in the VPN concentrator overwrites this setting.</td>
</tr>
<tr>
<td></td>
<td>Default: 60 seconds</td>
</tr>
<tr>
<td></td>
<td>Minimum: 0</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>Default: 30 seconds</td>
</tr>
<tr>
<td></td>
<td>Minimum: 0</td>
</tr>
<tr>
<td></td>
<td>Maximum: 600 seconds</td>
</tr>
</tbody>
</table>
From the drop-down list, choose the client authentication method:

- User and password
- Password only
- Certificate (LSC or MIC)

Default: User And Password

When True, a user password gets saved in the phone until either a failed login attempt occurs, a user manually clears the password, or the phone resets or loses power.

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

Before You Begin

Configure VPN Profile, on page 73

Procedure

Step 1 Choose Device > Device Settings > Common Phone Profile. The Find and List Common Phone Profiles window appears.
Step 2 Choose the search criteria to use.
Step 3 Click Find. The window displays a list of common phone profiles that match the search criteria.
Step 4 Click the common phone profile to which you want to add the VPN details. The Common Phone Profile Configuration window appears.
Step 5 In the VPN Information section, choose the appropriate VPN Group and VPN Profile.
Step 6 Click Save.
Step 7 Click Apply Config. The Apply Configuration window appears.
Step 8 Click Ok.
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 the Cisco Unified IP Phone Administration Guide for your Cisco Unified IP Phone model.

• Using a supported Cisco Unified IP Phone, establish the VPN connection.
Part IV

Monitoring and Recording

- Silent Monitoring, page 81
- Recording, page 89
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 via 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 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 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 enabled, the supervisor phone must also have encryption enabled in order to allow secure silent monitoring. If the agent phone has encryption enabled, but the supervisor phone does not, the monitoring request fails.

**Whisper Coaching**

Cisco 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 Cisco 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:


**Configure Silent Monitoring Task Flow**

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

**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 7</td>
<td>Determine which phones support silent monitoring by running a phone feature list report.</td>
</tr>
</tbody>
</table>
| Step 2 | Perform one of the following procedures:  
- Enable Built in Bridge for Phones Clusterwide, on page 83  
- Enable Built in Bridge for a Phone, on page 83 | 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 3 | Enable Monitoring Privileges for Supervisor, on page 84 | Add the supervisor to a group that allows silent monitoring. |
### Command or Action | Purpose
---|---
**Step 4** | Assign a Monitoring Calling Search Space, on page 85 | Set up the monitoring calling search space for the supervisor phone.
**Step 5** | Configure Silent Monitoring Notification Tones, on page 85 | Configure whether you want to play notification tones to the call participants.
**Step 6** | Configure Secure Silent Monitoring, on page 86 | Optional. If your calls are encrypted, configure secure silent monitoring.
**Step 7** | Cisco Unified Contact Center Express Configuration Task Flow | Configure silent monitoring in Cisco Unified Contact Center Express.

---

#### Enable Built in Bridge for Phones Clusterwide

When you set the Built-in-Bridge clusterwide service parameter to enabled, 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. In 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**.

**What to Do Next**

If you want to set the Built in Bridge for an individual phone, refer to **Enable Built in Bridge for a Phone**, on page 83
Otherwise, see **Enable Monitoring Privileges for Supervisor**, on page 84

#### 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 83.
Procedure

Step 1  In Cisco Unified CM Administration, choose **Device > Phone**.
Step 2  Click **Find**.
Step 3  Select the agent phone.
Step 4  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 5  Click **Save**.

What to Do Next

Enable Monitoring Privileges for Supervisor, on page 84

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**
  • **Enable Built in Bridge for Individual Phones**

Procedure

Step 1  In 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**.

What to Do Next

Assign a Monitoring Calling Search Space, on page 85
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.

Before You Begin
Enable Monitoring Privileges for Supervisor, on page 84

Procedure

Step 1 In 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 For each of the supervisor's phone lines that will be used for monitoring, perform the following steps:
   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.

What to Do Next
Configure Silent Monitoring Notification Tones, on page 85

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, Cisco Unified Communications Manager does not play notification tones. You must configure a service parameter to allow notification tones.

Before You Begin
Assign a Monitoring Calling Search Space, on page 85

Procedure

Step 1 In Cisco Unified CM Administration, choose System > Service Parameters.
Step 2 From the Server drop-down list box, choose the server one 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**  
If you changed the service parameter configuration, reset the agent phone.

---

**What to Do Next**

If you want to monitor secure calls that use sRTP, refer to the following procedure:

*Configure Secure Silent Monitoring, on page 86*

---

**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 86</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Assign Security Profile to Phone, on page 87</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**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose System &gt; Security &gt; Phone Security Profile.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Perform either of the following steps:</td>
</tr>
<tr>
<td></td>
<td>• Click <strong>Add New</strong> to create a new phone security profile.</td>
</tr>
</tbody>
</table>
• 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 box.

**Step 4**
Enter a **Name** for the Phone Security Profile.

**Step 5**
From the **Device Security Mode** drop-down list box, 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.

---

**What to Do Next**

Assign Security Profile to Phone, on page 87

---

**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.

**Before You Begin**

Configure an Encrypted Phone Security Profile, on page 86

---

**Procedure**

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

**Step 2**
Click **Find** and select the agent phone on which you want to configure a phone security profile.

**Step 3**
From the **Device Security Profile** drop-down list box, choose the phone security profile that you have set up.

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

**Step 5**
Repeat the previous steps for the supervisor phone.

---

**Silent Monitoring Interactions and Restrictions**

**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, Cisco Unified Communications Manager also puts the monitoring call into call preservation mode.</td>
</tr>
<tr>
<td>Transfer of secure monitoring call</td>
<td>Cisco 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>
</tbody>
</table>
Silent Monitoring Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<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>
</tbody>
</table>
| Secure Tones            | 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. |

Silent Monitoring Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Barge</td>
<td>Cisco 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>Cisco Unified Communications Manager does not support transferring Secure Silent Monitoring calls over an intercluster trunk.</td>
</tr>
</tbody>
</table>
CHAPTER 11

Recording

• Recording Overview, page 89
• Recording Prerequisites, page 91
• Recording Configuration Task Flow, page 91
• Recording Interactions and Restrictions, page 104

Recording Overview

Call recording is a Cisco 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 customer RTP 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.

Cisco 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 Cisco Unified Communications Manager over a SIP trunk.

Cisco 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. Cisco 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 via 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.
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, Cisco Unified Communications Manager may not select your preferred choice as the recording media source. The following table displays the logic Cisco Unified Communications Manager uses to select the recording media source.

Table 2: 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 Cisco Unified Communications Manager selects is unavailable, Cisco Unified Communications Manager attempts to use an alternate source. The following table shows the logic Cisco Unified Communications Manager uses to select an alternate source for recording media.

Table 3: 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 IP Phone support—To view a list of the Cisco Unified IP phones 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 7.

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

Recording 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>Create a Recording Profile, on page 92</td>
<td>Create a recording profile.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Optional. Configure SIP Profile with Conference Bridge Identifier for Recording, on page 92</td>
<td>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 93</td>
<td>Configure the recorder as a SIP trunk device.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configure Route Pattern for Recording, on page 94</td>
<td>Create a route pattern that routes to the recorder.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Configure Agent Phone Line for Recording, on page 94</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. Complete 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 95</td>
<td><strong>Note</strong> 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 95</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Enable Gateway for Recording, on page 96</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 97</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>Optional. Configure Recording Redundancy Using Route Groups, on page 97</td>
<td>Configure redundancy for your recording servers. There are many methods for configuring redundancy for your recording servers. For the preferred redundancy method for your deployment, refer to your vendor.</td>
</tr>
</tbody>
</table>
Create a Recording Profile

Procedure

Step 1 In 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 recorder that contains the partition with the route pattern that is configured for the recording server.
Step 6 Click Save.

What to Do Next

Configure SIP Profile with Conference Bridge Identifier for Recording, on page 92

Configure SIP Profile with Conference Bridge Identifier for Recording

If you want Cisco Unified Communications Manager to deliver the conference bridge identifier to the recorder, configure the SIP Profile by performing the following steps:

Before You Begin

Create a Recording Profile, on page 92
Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Device Settings > SIP Profiles.
Step 2 Select the SIP profile that you want to use for your network.
Step 3 Check the Deliver Conference Bridge Identifier check box.
Step 4 Click Save.

What to Do Next
Configure SIP Trunks for Recording, on page 93

Configure SIP Trunks for Recording

Assign the recording server information in the SIP Trunk Configuration window by performing the following steps:

Before You Begin
Configure SIP Profile with Conference Bridge Identifier for Recording, on page 92

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Trunk.
Step 2 From the Trunk Type drop-down list, choose SIP Trunk.
Step 3 From the Protocol drop-down list, choose None.
Step 4 From the SIP Profile drop-down list, select the SIP profile that you want to use in your network.
Step 5 In the Destination Address field of the SIP Information pane, enter an IP address, fully qualified domain name, or DNS SRV record for the recording server.
Step 6 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 7 Click Save.

What to Do Next
Perform one of the following procedures, depending on whether you are configuring your main recording server, or multiple recording servers for redundancy:

- Configure Route Pattern for Recording, on page 94
- Add Recording Servers to Route Group, on page 98
Configure Route Pattern for Recording

This procedure describes the route pattern configurations that are specific to recorders. You must configure a route pattern that routes to the recording server.

Before You Begin
Configure SIP Trunks for Recording, on page 93

Procedure

Step 1 In 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 detailed field descriptions, refer to the online help system.
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.

What to Do Next
Configure Agent Phone Line for Recording, on page 94

Configure Agent Phone Line for Recording

Before You Begin
Configure Route Pattern for Recording, on page 94

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Find.
Step 3 Select the agent's phone.
The Phone Configuration window appears.
Step 4 In the left Association pane, click the phone line for the agent.
The Directory Number Configuration window opens, displaying settings for the agent's phone line.
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**
Click **Save**.

---

**What to Do Next**
Enable the Built in Bridge on the phone by performing one of the following procedures:

- Enable Built in Bridge for Cluster, on page 95
- Enable Built in Bridge for a Phone, on page 95

---

**Enable Built in Bridge for Cluster**

When you set the Built-in-Bridge clusterwide service parameter to enabled, 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.

**Before You Begin**
Configure Agent Phone Line for Recording, on page 94

**Procedure**

**Step 1**
In 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 **Built in Bridge Enable** service parameter to **On**.

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

**What to Do Next**
If you want to configure the Built in Bridge on an individual phone, see Enable Built in Bridge for a Phone, on page 95

Otherwise, see Enable Gateway for Recording, on page 96

---

**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.
Before You Begin

Refer to Configure Agent Phone Line for Recording, on page 94

Optionally, use a service parameter to set the Built in Bridge defaults across the cluster. For details, see Enable Built in Bridge for Cluster, on page 95

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Find.
Step 3 Select the agent phone.
Step 4 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 5 Click Save.

What to Do Next

Enable Gateway for Recording, on page 96

Enable Gateway for Recording

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.

Before You Begin

Enable Built in Bridge for a Phone, on page 95

Procedure

Step 1 Configure Cisco 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:

What to Do Next

Configure Recording Notification Tones, on page 97

**Configure Recording Notification Tones**

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.

**Before You Begin**

Enable Gateway for Recording, on page 96

**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 SIP Trunks for Recording, on page 93</td>
<td>Configure a separate SIP trunk for each recording server.</td>
</tr>
</tbody>
</table>

What to Do Next

Optional. Configure Recording Redundancy Using Route Groups, on page 97

**Configure Recording Redundancy Using Route Groups**

This task flow describes how to configure recording redundancy using route groups. For the preferred method of configuring recording redundancy for your deployment, refer to your vendor.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configure SIP Trunks for Recording, on page 93</td>
<td>Configure a separate SIP trunk for each recording server.</td>
</tr>
</tbody>
</table>
### Command or Action | Purpose
--- | ---
**Step 2** | **Add Recording Servers to Route Group, on page 98** | Add your recording SIP trunks to a route group.
**Step 3** | **Add Route Group to Route List, on page 98** | Add your recording route group to a route list.
**Step 4** | **Configure Route Pattern for Recording, on page 94** | Configure a route pattern that routes recording calls to your route list.

---

### Add Recording Servers to Route Group

Configure SIP Trunks for Recording, on page 93

**Procedure**

1. **Step 1** In Cisco Unified CM Administration, choose **Call Route > Route Hunt/List > Route Group**.
2. **Step 2** Perform one of the following steps:
   - Click **Find** and select an existing route group.
   - Click **Add New** to create a new route group.
3. **Step 3** Complete the fields in the Route Group Configuration window. See the online help for more information on the fields and their options.
4. **Step 4** In the Available Devices pane, for each SIP trunk on which you configured a recording server, select the SIP trunk and click **Add to Route Group** until each recording server appears in the Selected Devices pane.
5. **Step 5** Use the up and down arrows to adjust the priority setting for each recording server.
6. **Step 6** Click **Save**.

**What to Do Next**

Add Route Group to Route List, on page 98

### Add Route Group to Route List

**Before You Begin**

Add Recording Servers to Route Group, on page 98

**Procedure**

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

**Step 3**  Click **Add Route Group**.
The Route List Details Configuration window opens.

**Step 4**  From the Route Group drop-down list, select the route group that you created for your recording servers.

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

**Step 6**  Complete the remaining fields in the Route List Configuration window. See the online help for more information on the fields.

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

---

**What to Do Next**

*Configure Route Pattern for Recording*, on page 94

---

**Configure a Record Feature Button**

If your phone uses feature buttons, use the following procedure to assign the Record feature button to your phone.

**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 Phone Button Template for Recording, on page 99</td>
<td>Configure a phone button template that includes the Record button.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Phone Button Template with a Phone, on page 100</td>
<td>Associate the phone button template that you set up for recording to the phone.</td>
</tr>
</tbody>
</table>

---

**Configure a Phone Button Template for Recording**

Create a phone button template that includes the Record feature button.

**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>From Cisco Unified CM Administration, choose <strong>Device &gt; Device Settings &gt; Phone Button Template</strong>. The Find and List Phone Button Templates window appears.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Click <strong>Find</strong>. The window displays a list of templates for the supported phones.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Perform this step 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 <strong>Copy</strong>. b) In the <strong>Phone Button Template Information</strong> field, enter a new name for the template.</td>
<td></td>
</tr>
</tbody>
</table>
c) Click Save.

**Step 4** Perform this step if you want to add phone buttons to an existing template.
   a) Enter search criteria and click **Find**.
   b) Choose an existing template.
      The **Phone Button Template Configuration** window appears.
**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. See the What to Do Next section for more information.

---

**What to Do Next**

*Associate a Phone Button Template with a Phone,* on page 100

**Associate a Phone Button Template with a Phone**

Associate the phone button template that you created for the Record button to the phone.

**Before You Begin**

*Configure a Phone Button Template for Recording,* on page 99

**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>Device &gt; Phone</strong>. The <strong>Find and List Phones</strong> window is displayed.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>From the <strong>Find and List Phones</strong> window, click <strong>Find</strong>. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Choose the phone to which you want to add the phone button template. The <strong>Phone Configuration</strong> window appears.</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>
Configure a Record Softkey

If your phone uses softkeys, perform the following tasks to add a Record softkey to the phone. 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>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure a softkey template that includes the Record softkey.</td>
</tr>
<tr>
<td><strong>Step 2</strong></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>
</tbody>
</table>

Configure a Softkey Template for Recording

**Procedure**

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

**Step 2**
Perform this step 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) In the Softkey Template Name field, enter a new name for the template.
- d) Click Save.

**Step 3**
Perform this step to add softkeys to an existing template.
- a) Enter search criteria and click Find.
- b) Choose an existing template.
  The Softkey Template Configuration window appears.

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

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. See the What to Do Next section for more information.

What to Do Next

Perform either of the following tasks:

- Associate a Softkey Template with a Phone, on page 102
- Associate a Softkey Template with a Common Device Configuration, on page 103

Associate a Softkey Template with a Phone

You can assign the Record softkey to the phone by associating the softkey template that includes the Record softkey directly to a phone.

Before You Begin

Configure a Softkey Template for Recording, on page 101

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.

Step 2 Choose the phone to which you want to add the softkey template. The Phone Configuration window appears.

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

This task flow describes how to add a Record softkey to the phone by associating the softkey template to a Common Device Configuration.

Before You Begin
Configure a Softkey Template for Recording, on page 101

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 186</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Common Device Configuration with a Phone, on page 187</td>
</tr>
</tbody>
</table>

Add a Softkey Template to the Common Device Configuration

Before You Begin
Configure a Softkey Template for Recording, on page 101

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Common Device Configuration. The Find and List Common Device Configuration window appears.

Step 2 Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
   c) Click Save.

Step 3 Perform this step to add the softkey template to an existing Common Device Configuration.
   a) Enter search criteria and click Find.
   b) Choose an existing Common Device Configuration. The Common Device Configuration window appears.

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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.
   • If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.
What to Do Next
Add Common Device Configuration to Phone, on page 104

Add Common Device Configuration to Phone

Before You Begin
Associate a Softkey Template with a Common Device Configuration, on page 103

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.
Step 2 Find the phone to which 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.

Recording Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interactions</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>Secure Tones</td>
<td>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.</td>
</tr>
<tr>
<td>Customer Voice Portal</td>
<td>Agent - customer calls that are routed through the Customer Voice Portal may be recorded using the agent phone as the recording source.</td>
</tr>
<tr>
<td>SIP Proxy Servers</td>
<td>If you are using the gateway as your recording source, you cannot place SIP proxy servers between Cisco Unified Communications Manager and the gateway.</td>
</tr>
<tr>
<td>Busy Hour Call Completion Rate</td>
<td>Each recording session adds two calls to the Busy Hour Call Completion (BHCC) rate with a minimal impact on CTI resources.</td>
</tr>
</tbody>
</table>
Part V

Call Center Features

- Agent Greeting, page 107
- Auto-Attendant, page 111
- Manager Assistant, page 121
Agent Greeting

Agent Greeting Overview

Agent Greeting enables Cisco 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


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...

**Before You Begin**
Ensure that you enable Built In Bridge. To view the details, see Configure Built In Bridge, on page 109.

**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 media server for Agent Greeting.</td>
<td>Agent Greeting uses the Unified CVP media server to store and serve prompt and greeting files.</td>
</tr>
<tr>
<td></td>
<td>• Configure a server to act as a media server.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Add the media server in Unified CVP.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure the media server to write files.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Republish .tcl scripts to Voice Extensible Markup Language (VXML) Gateway.</td>
<td>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.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Set the cache size on the VXML Gateway.</td>
<td>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.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Create voice prompts to record greetings.</td>
<td>Create audio files for each of the voice prompts that agents hear as they record a greeting.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure call types.</td>
<td>Complete to record and play agent greetings.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Configure a dialed number.</td>
<td>Complete to record and play agent greetings.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Schedule the script.</td>
<td></td>
</tr>
<tr>
<td>Step 8</td>
<td>Define network VRU scripts.</td>
<td>For Agent Greeting record and play scripts to interact with Unified CVP, Network VRU scripts are required.</td>
</tr>
<tr>
<td>Step 9</td>
<td>(Optional) Import sample Agent Greeting scripts.</td>
<td></td>
</tr>
</tbody>
</table>
Modify the Unified CCE call routing scripts to use the Play Agent Greeting script.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 10</strong></td>
<td>Modify the Unified CCE call routing scripts.</td>
</tr>
</tbody>
</table>

## 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

1. **Step 1** In Cisco Unified CM Administration, choose **Device > Phone**.
2. **Step 2** Click **Find**.
3. **Step 3** Select the agent phone.
4. **Step 4** 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.
5. **Step 5** Click **Save**.

## 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 Cisco 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 upon the callers DTMF input choices.


Cisco Unity Connection Configuration Task Flow

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

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure CTI Route Point, on page 113</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>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Auto-Attendant System Call Handler, on page 114</td>
<td>Call handlers answer calls, greet callers with recorded prompts, provide callers with information and options, route calls, and take messages. <strong>Note</strong> You can customize the greeting for the AutoAttendant Call Handler by choosing Edit &gt; Greetings. For more information about customizing greetings, see the System Administration Guide for Cisco Unity Connection at <a href="http://www.cisco.com/c/en/us/support/unified-communications/unity-connection/products-maintenance-guides-list.html">http://www.cisco.com/c/en/us/support/unified-communications/unity-connection/products-maintenance-guides-list.html</a>.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Configure Caller Input Option, on page 114</td>
<td>Caller input options 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.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure Extension for Operator Call Handler, on page 115</td>
<td>Configure an extension for the operator to allow callers to speak to an operator during a call handler greeting.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Modify Standard Call Transfer Rule for Operator, on page 116</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Step 6</strong> Update Default System Transfer Restriction Table, on page 116</td>
<td>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.</td>
</tr>
</tbody>
</table>

**Configure CTI Route Point**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > CTI Route Point. The Find and List CTI Route Points window is displayed.

**Step 2** Click Add New. The CTI Route Point Configuration window is displayed.

**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.

**Step 9**
From the Call Forward and Call Pickup Settings area, for Forward All, choose the appropriate calling search space and check the Voice Mail check box.

**Step 10**
Click Save.

---

**Configure Auto-Attendant System Call Handler**

**Before You Begin**

Ensure that you Configure CTI Route Point, on page 113

**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.
The Search Call Handlers window is displayed.

**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.

---

**What to Do Next**

Configure Caller Input Option, on page 114

---

**Configure Caller Input Option**

**Before You Begin**

Ensure that you:

- Configure CTI Route Point, on page 113
- Configure Auto-Attendant System Call Handler, on page 114
**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**. The **Search Call Handlers** window is displayed.

**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.

**What to Do Next**

*Configure Extension for Operator Call Handler*, on page 115

**Configure Extension for Operator Call Handler**

**Before You Begin**

Ensure that you:

- Configure CTI Route Point, on page 113
- Configure Auto-Attendant System Call Handler, on page 114
- Configure Caller Input Option, on page 114

**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**. The **Search Call Handlers** window is displayed.

**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.

---

**What to Do Next**

Modify Standard Call Transfer Rule for Operator, on page 116

---

**Modify Standard Call Transfer Rule for Operator**

**Before You Begin**

Ensure that you:

- Configure CTI Route Point, on page 113
- Configure Auto-Attendant System Call Handler, on page 114
- Configure Caller Input Option, on page 114
- Configure Extension for Operator Call Handler, on page 115

**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**. The **Search Call Handlers** window is displayed.

**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**.

---

**What to Do Next**

Update Default System Transfer Restriction Table, on page 116

---

**Update Default System Transfer Restriction Table**

**Before You Begin**

Ensure that you:

- Configure CTI Route Point, on page 113
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. The Search Restriction Tables window is displayed.

Step 2  Click Default System Transfer. The Edit Restriction Table Basics (Default System Transfer) window is displayed.

Step 3  Uncheck the checkbox 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 Cisco Unified Communications Manager, see http://www.cisco.com/en/US/docs/voice_ip_comm/uc_system/unified/communications/system/versions/IPTMtrix.html.

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 Cisco Unified Communications Manager.

Cisco Unified CCX Auto-Attendant 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 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>Step 2</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. Caution: All media termination strings begin with &quot;auto&quot; 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.</td>
</tr>
<tr>
<td>Step 3</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>Step 4</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>
</tbody>
</table>
Command or Action | Purpose
---|---
**Step 5** Customize Auto-Attendant.  
- Modify an existing Auto-Attendant instance  
- Configure the Auto-Attendant prompts | The Cisco Unified CCX Administration page allows you to modify any existing Auto-Attendant instance as necessary.

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.

**Cisco Unified CCX Auto-Attendant Troubleshooting**


**Cisco Unity Express Configuration**


**Cisco Unity Express Auto-Attendant Troubleshooting**

Manager Assistant

- Cisco Unified Communications Manager Assistant Overview, page 121
- Manager Assistant Prerequisites, page 123
- Manager Assistant Task Flow for Proxy Lines, page 123
- Manager Assistant Task Flow for Shared Lines, page 132
- Manager Assistant Interactions and Restrictions, page 151
- Cisco Unified Communications Manager Assistant Troubleshooting, page 154

Cisco Unified Communications Manager Assistant Overview

The Cisco 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 Cisco 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 Cisco 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 about the Manager Assistant, see the *Cisco Unified Communications Manager Assistant User 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 with the exception of Manager Configuration. Manager Assistant automatically logs a manager into the Cisco IP Manager Assistant service when the Cisco IP Manager Assistant service starts.

  Note  
  Managers also have access to Cisco 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 about the Manager Assistant, see the *Cisco Unified Communications Manager Assistant User 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:
  - Cisco 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, 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 JRE1.4.2_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 Cisco Unified Communications Manager Bulk Administration Tool. For more information, see the Cisco Unified Communications Manager Bulk Administration Guide.

Manager Assistant Task Flow for Proxy Lines

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Run the Cisco Unified CM Assistant Configuration Wizard, on page 124</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Manager And Assign Assistant For Proxy Line, on page 130</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure Assistant Line Appearances for Proxy Line, on page 131</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Install Assistant Console Plugin, on page 150</td>
<td>The assistant accesses the Cisco Unified Communications Manager Assistant features by using the Assistant Console application and the Cisco</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 the Cisco Unified Communications Manager Bulk Administration Guide.

Before You Begin

Ensure that the configuration wizard runs on the same server (the Cisco Unified Communications Manager server) as the Bulk Administration Tool.

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
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 tomeat/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 the **Manager Assistant Service Parameters for Proxy Line**, on page 126.
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.

<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.</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>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 <strong>IPMASEcureSysUser</strong> that this Manager Assistant will use to open a secure connection to CTIManager.</td>
</tr>
<tr>
<td></td>
<td>Configure this parameter if <strong>CTIManager Connection Security Flag</strong> is enabled.</td>
</tr>
<tr>
<td><strong>Clusterwide Parameters (Parameters that apply to all servers)</strong></td>
<td></td>
</tr>
<tr>
<td>Important</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>Default value: 2912</td>
</tr>
<tr>
<td>Setting</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------------------------</td>
<td>----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Cisco IPMA Assistant Console Heartbeat</td>
<td>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</td>
</tr>
<tr>
<td>Console Request Timeout</td>
<td>This parameter specifies the time, in seconds, that the IPMA Assistant Consoles wait to receive a response from the Cisco IPMA server.</td>
</tr>
<tr>
<td>Cisco IPMA RNA Forward Calls</td>
<td>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</td>
</tr>
<tr>
<td>Alpha Numeric UserID</td>
<td>This parameter determines whether Cisco IPMA Assistant Phone uses an alphanumeric user ID or a numeric user ID.</td>
</tr>
<tr>
<td>Cisco IPMA RNA Timeout</td>
<td>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</td>
</tr>
<tr>
<td>CTIManager Connection Security Flag</td>
<td>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.</td>
</tr>
<tr>
<td>Redirect call to Manager upon failure to reach Assistant</td>
<td>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</td>
</tr>
</tbody>
</table>
### Setting | Description
--- | ---
**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>
<tbody>
<tr>
<td>Enable Multiple Active Mode</td>
<td>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</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pool 2: 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 second instance of Cisco IPMA. Configure the Cisco IP Manager Assistant service parameters on this node.</td>
</tr>
</tbody>
</table>

<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>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<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>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<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.

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Assistant Softkey Template</td>
<td>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.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Manager Softkey Template for Proxy Mode</td>
<td>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.</td>
</tr>
</tbody>
</table>

### Clusterwide Parameters (IPMA Device Configuration Defaults for Proxy Mode)
<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Manager Partition</td>
<td>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 Cisco IPMA Configuration Wizard is run, it will populate this value. This parameter applies only for managers that use proxy mode.</td>
</tr>
<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 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>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 the 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 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>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 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><strong>Clusterwide Parameters (Proxy Directory Number Range for Proxy Mode)</strong></td>
<td></td>
</tr>
<tr>
<td>Starting Directory Number</td>
<td>This parameter specifies the starting directory number that is used as the starting number for automatic generation of proxy directory numbers during IPMA assistant configuration. After an auto-generated proxy line number is used for an assistant, the next number will be generated for the next assistant, and so on. This parameter applies only for assistants that use proxy mode.</td>
</tr>
<tr>
<td>Ending Directory Number</td>
<td>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.</td>
</tr>
</tbody>
</table>
### Clusterwide Parameters (Proxy Directory Number Range for Proxy Mode)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Characters to be Stripped from Manager DN</td>
<td>This parameter specifies the number of characters to be stripped from the manager directory number (DN) in the process of generating the proxy DN. Generating a proxy DN may involve stripping some number of digits and adding a prefix. Digits are stripped starting from the left. This parameter applies only for assistants that use proxy mode.</td>
</tr>
<tr>
<td>Prefix for Manager DN</td>
<td>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.</td>
</tr>
</tbody>
</table>

### Configure Manager And Assign Assistant For Proxy Line

For information about configuring a new user and associating a device to the user, see [http://www.cisco.com/e/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmefg/CUCM_BK_C95ABA82_00_admin-guide-100.html](http://www.cisco.com/e/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmefg/CUCM_BK_C95ABA82_00_admin-guide-100.html).

**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. The Find and List Users window is displayed.

**Step 2** Click Find.

The search result displays all the end users that are configured in Cisco 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 151.

**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 Cisco 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.

Note  Make sure you configure manager information and assign an assistant to the manager before you configure assistant information for an assistant.

Note  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. The Find and List Users window is displayed.

Step 2  Click Find. The search result displays all the end users that are configured in Cisco Unified Communications Manager.

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. 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.
The Assistant Configuration window is displayed.

**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**

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>Configure Partitions for Manager Assistant Shared Line Support, on page 133</strong></td>
<td>Configure a partition for lines that is used by Manager Assistant.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>Configure Calling Search Spaces for Manager Assistant Shared Line Support, on page 135</strong></td>
<td>Configure calling search spaces for manager and assistant lines.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>Configure Cisco IP Manager Assistant Service Parameters, on page 135</strong></td>
<td>Configure these parameters to use automatic configuration for managers and assistants.</td>
</tr>
</tbody>
</table>
| **Step 4** | *(Optional) Configure Intercom Settings*  
  - **Configure an Intercom Partition, on page 136**  
  - **Configure an Intercom Calling Search Space, on page 284**  
  - **Configure an Intercom Directory Number, on page 290**  
  - **Configure an Intercom Translation Pattern, on page 285** | Configure intercom settings for managers and assistants. |
<p>| <strong>Step 5</strong> | <em>(Optional) Configure Multiple Manager Assistant Pool, on page 138</em> | Configure multiple pools if you need to support a large number of managers and assistants. You can configure up to three active Cisco IP pools. |</p>
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Manager Assistant servers, with each managing up to 2500 pairs of managers and assistants.</td>
</tr>
</tbody>
</table>
| **Step 6**       | (Optional) **Configure Secure TLS Connection to CTI for Manager Assistant**  
|                   |  
|                   | • Configure IPMASecureSysUser Application User, on page 140  
|                   | • Configure CAPF Profile, on page 140  
|                   | • Configure Cisco IP Manager Assistant, on page 142  
|                   | Follow these procedures if your system is running in mixed mode. |
| **Step 7**       | **Configure CTI Route Point, on page 142**  
|                   | Cisco Unified Communications Manager Assistant requires creation of CTI route point to intercept and route calls from managers. |
| **Step 8**       | **Configure IP Phone Services for Manager and Assistant, on page 143**  
| **Step 9**       | **Configure Phone Button Templates for Manager, Assistant, and Everyone, on page 146**  
| **Step 10**      | **Configure Manager and Assign Assistant for Shared Line Mode, on page 148**  
| **Step 11**      | **Configure Assistant Line Appearances for Shared Line, on page 149**  
| **Step 12**      | **Install Assistant Console Plugin, on page 150**  
|                   | The assistant accesses the Cisco Unified Communications Manager Assistant features by using the Assistant Console application and the Cisco Unified IP Phone. The Assistant Console provides call-control functions such as answer, divert, transfer, and hold. |
| **Step 13**      | **Configure the manager and assistant console applications.**  

**Configure Partitions for Manager Assistant Shared Line Support**

You must create three partitions: Generated_Everyone, Generated_Managers, and Generated_Route_Point.
Procedure

Step 1  In Cisco Unified Communications Manager 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 Related Topics section 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 timeschedule to associate with this partition. The timeschedule 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.

Related Topics

**Partition Name Guidelines for Manager Assistant Shared Line Support,** on page 134

**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>170</td>
</tr>
<tr>
<td>3 characters</td>
<td>128</td>
</tr>
<tr>
<td>4 characters</td>
<td>102</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, select **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**.

---

### 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.

<table>
<thead>
<tr>
<th>Partition Name Length</th>
<th>Maximum Number of Partitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>5 characters</td>
<td>86</td>
</tr>
<tr>
<td></td>
<td>...</td>
</tr>
<tr>
<td>10 characters</td>
<td>46</td>
</tr>
<tr>
<td>15 characters</td>
<td>32</td>
</tr>
</tbody>
</table>
Procedure

Step 1  From Cisco Unified CM Administration, choose System > Service Parameters. The Service Parameter Configuration window is displayed.

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 136</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure an Intercom Calling Search Space, on page 137</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure an Intercom Directory Number, on page 137</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure an Intercom Translation Pattern, on page 138</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 the Add New button. 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, Cisco 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 Related Topics section 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 Related Topics section 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 the Add New button.  
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. See the Related Topics section for more information about the fields and their configuration options.

**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 the **Copy** button beside 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. See the Related Topics section for more information about the fields and their configuration options.

**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.

---

### Configure Multiple Manager Assistant Pool

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **System > Service Parameters**. The **Service Parameter Configuration** window is displayed.

**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.
The Service Parameter Configuration window, which lists the parameters, is displayed.

**Step 4**
Click Advanced.
The advanced parameters for Clusterwide Parameters (Parameters that apply to all servers) are displayed.

**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 WD SecureSysUser application user credentials to establish a secure TLS connection to CTI to make calls.

To configure the WD SecureSysUser 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 the Cisco Unified Communications Manager Security Guide.

- 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 the Cisco Unified Communications Manager Security Guide.

- Activate the Cisco Certificate Authority Proxy Function (CAPF) service on the first node.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure IPMASecureSysUser Application User, on page 140</td>
<td>Configure IPMASecureSysUser Application User.</td>
</tr>
</tbody>
</table>
### Configure IPMASecureSysUser Application User

Use this procedure to configure IPMASecureSysUser application user.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Find.</td>
<td>Configure service parameters for the Cisco IP Manager Assistant service.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Find and List Application Users Application window, choose WDSecureSysUser.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure the fields in the Application User Configuration window and click Save.</td>
<td></td>
</tr>
</tbody>
</table>

### 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**

<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 User Management &gt; Application User CAPF Profile.</td>
<td>Configure Certificate Authority Proxy Function (CAPF) Profile for the IPMASecureSysUser Application User.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Perform one of the following tasks:</td>
<td>Configure service parameters for the Cisco IP Manager Assistant service.</td>
</tr>
<tr>
<td></td>
<td>• To add a new CAPF profile, click Add New in the Find window.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• To copy an existing profile, locate the appropriate profile and click the Copy icon for that record in the Copy column.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>To update an existing entry, locate and display the appropriate profile.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure or update the relevant CAPF profile fields. See the Related Topics section information about the fields and their configuration options.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Repeat the procedure for each application and end user that you want to use security.</td>
<td></td>
</tr>
</tbody>
</table>
**What to Do Next**

Configure Cisco IP Manager Assistant, on page 142

**Related Topics**

CAPF Profile Settings, on page 141

### 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 <strong>End User CAPF Profile</strong> 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 <strong>Application User CAPF Profile</strong> 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>• <strong>No Pending Operation</strong>—This message is displayed when no certificate operation is occurring. (default setting)</td>
</tr>
<tr>
<td></td>
<td>• <strong>Install/Upgrade</strong>—This option installs a new certificate or upgrades an existing locally significant certificate for the application.</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 <strong>JTAPI/TSP Preferences</strong> 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.</td>
</tr>
<tr>
<td></td>
<td>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 <strong>Authentication String</strong> field.</td>
</tr>
</tbody>
</table>
**Setting** | **Description**
--- | ---
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 IP Manager Assistant**

**Procedure**

**Step 1**  
In 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. 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.

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

**Step 6**  
Repeat the procedure on each server on which the service is active.

---

**Configure CTI Route Point**

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **Device > CTI Route Point**. The **Find and List CTI Route Points** window is displayed.

**Step 2**  
Click **Add New**.
The **CTI Route Point Configuration** window is displayed.

**Step 3**  
In the **Device Name** field, enter the device name.

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

**Step 5**  
From the **Calling Search Space** drop-down list, choose **Generated_CSS_M_E**.

**Step 6**  
Check the **Use Device Pool Calling Party Transformation CSS** check box.

**Step 7**  
Click **Save**.  
The **Add successful** status message is displayed.

**Step 8**  
From the Association area, click **Line [1] - Add a new DN**.  
The **Directory Number Configuration** window is displayed.

**Step 9**  
Enter a directory number in the **Directory Number** field.

**Step 10**  
From the **Route Partition** drop-down list, choose **Generated_Route_Point**.

**Step 11**  
Click **Save**.

---

**Configure IP Phone Services for Manager and Assistant**

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **Device > Device Settings > Phone Services**.  
The **Find and List IP Phone Services** window is displayed.

**Step 2**  
Click **Add New**.  
The **IP Phone Services Configuration** window is displayed.

**Step 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 143

---

**Cisco IP Phone Services Configuration Fields**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Service Information</td>
<td></td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</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. Note: Cisco 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. 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>
<tr>
<td>Service URL</td>
<td>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 Cisco Unified Communications Manager cluster. Do not specify a Cisco Unified Communications Manager server or any server that is associated with Cisco Unified Communications Manager (such as a TFTP server or directory database publisher server). For the services to be available, the phones in the Cisco 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/&lt;name of service&gt; 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.</td>
</tr>
</tbody>
</table>
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 Cisco Unified Communications Manager cluster. Do not specify a Cisco Unified Communications Manager server or any server that is associated with Cisco Unified Communications Manager (such as a TFTP server or publisher database server).

For the services to be available, the phones in the Cisco 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.

**Service Vendor**  
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.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure-Service URL</td>
<td>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 Cisco Unified Communications Manager cluster. Do not specify a Cisco Unified Communications Manager server or any server that is associated with Cisco Unified Communications Manager (such as a TFTP server or publisher database server). For the services to be available, the phones in the Cisco Unified Communications Manager cluster must have network connectivity to the server. <strong>Note</strong> 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.</td>
</tr>
<tr>
<td>Service Category</td>
<td>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.</td>
</tr>
<tr>
<td>Service Type</td>
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</tr>
<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>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Service Version</td>
<td>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:</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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 Cisco 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.</td>
</tr>
<tr>
<td></td>
<td>This field displays as blank for Cisco-provided default services. You can enter numbers and periods in this field (up to 16 ASCII characters).</td>
</tr>
<tr>
<td>Enable</td>
<td>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).</td>
</tr>
<tr>
<td></td>
<td>Uncheck the check box to remove the service from the phone configuration file and the phone.</td>
</tr>
</tbody>
</table>

**Service Parameter Information**

<table>
<thead>
<tr>
<th>Parameters</th>
<th>This pane lists the service parameters that apply to this IP phone service. Use the following buttons to configure service parameters for this pane:</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>• <strong>New Parameter</strong>—Click this button to display the Configure Cisco Unified IP Phone Service Parameter 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 Configure Cisco Unified IP Phone Service Parameter 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>Configure a Phone Button Template for Manager Assistant</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Manager Assistant Button Template with a Phone, on page 147</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>
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<td><strong>Step 1</strong></td>
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</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Manager Assistant Button Template with a Phone, on page 147</td>
</tr>
</tbody>
</table>

**Associate a Manager Assistant Button Template with a Phone**

Procedure

- **Step 1**  From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window is displayed.
- **Step 2**  From the **Find and List Phones** window, click **Find**. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.
- **Step 3**  Choose the phone to which you want to add the phone button template. The **Phone Configuration** window appears.
- **Step 4**  In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.
- **Step 5**  Click **Save**. A dialog box is displayed with a message to press **Reset** to update the phone settings.
Configure Manager and Assign Assistant for Shared Line Mode

Procedure

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

Step 2 Click Find. The search result displays all the end users that are configured in Cisco Unified Communications Manager.

Step 3 From the Related Links drop-down list, choose Manager Configuration and click Go. The Manager Configuration window is displayed.

Step 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 Cisco Unified Communications Manager Assistant intercom feature is used for the Cisco Unified IP Phones 7940 and 7960.

Step 5 Check Uses Shared Lines check box.

Step 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.

Note 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 Cisco 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.

Related Topics

Manager Assistant Interactions and Restrictions, on page 151
Configure Assistant Line Appearances for Shared Line

Administrators can set up one or more lines with a shared line appearance. The Cisco 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

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From Cisco Unified CM Administration, choose User Management &gt; End User. The Find and List Users window is displayed.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find. The search result displays all the end users that are configured in Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click on the username to display user information for the chosen assistant. The End User Configuration window is displayed.</td>
</tr>
<tr>
<td>Step 4</td>
<td>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.</td>
</tr>
<tr>
<td>Step 5</td>
<td>From the Device Name drop-down list, choose the device name to associate with the assistant.</td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Intercom Line drop-down list, choose the incoming intercom line appearance for the assistant.</td>
</tr>
</tbody>
</table>
| 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. 
*Note* 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 Cisco 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 and Restrictions

## 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 <strong>Cisco Unified CM Assistant Configuration Wizard</strong> creates for Cisco Unified IP Phones support only the Cisco Unified Communications Manager intercom lines. For more information, see the <em>Cisco Unified Communications Manager Bulk Administration Guide</em>.</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 <em>Extension Mobility</em>, on page 371.</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 Interactions and Restrictions

<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 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.</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: * Cisco Unified Serviceability Administration Guide * Cisco Unified Communications Manager CDR Analysis and Reporting Administration Guide</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 Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom</td>
<td>Manager Assistant supports the following two types of intercom:</td>
</tr>
<tr>
<td></td>
<td>• Manager Assistant intercom (used with Cisco Unified IP Phones 7940 and 7960). You can configure this intercom feature using the DN configuration and end user (manager and assistant) configuration windows.</td>
</tr>
<tr>
<td></td>
<td>• Cisco Unified Communications Manager intercom (used with Cisco Unified IP Phones 7940 and 7960). You can configure this intercom feature using the intercom partition, intercom calling search space, intercom directory number, intercom translation pattern, DN, and end user (manager and assistant) configuration windows.</td>
</tr>
<tr>
<td>Message Waiting Indicator</td>
<td>The Message Waiting Indicator feature interacts with proxy line support only.</td>
</tr>
<tr>
<td></td>
<td>The Message Waiting Indicator (MWI) on and off numbers should have the partition of the manager line in their calling search space. The partition can exist in any order of priority within each calling search space.</td>
</tr>
<tr>
<td>Time-of-Day Routing</td>
<td>The Time-of-Day feature interacts with proxy line support only.</td>
</tr>
<tr>
<td></td>
<td>Time-of-Day routing routes calls to different locations based on the time that the call gets made; for example, during business hours, calls get routed to a manager office, and after hours, the calls go directly to voicemail service.</td>
</tr>
<tr>
<td></td>
<td>For more information about Time-of-Day Routing, see the System Configuration Guide for Cisco Unified Communications Manager.</td>
</tr>
</tbody>
</table>

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>
<tr>
<td>Feature</td>
<td>Restriction</td>
</tr>
<tr>
<td>----------------</td>
<td>------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<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 Cisco Unified Communications Manager Assistant Intercom lines feature. Cisco Unified IP Phones 7900 (except 7940 and 7960) support only the Cisco 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 Cisco Unified Communications Manager–controlled line). Manager Assistant supports up to 3500 managers and 3500 assistants per Cisco 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 Cisco Unified Communications Manager Intercom feature. The system does not support calls between the Cisco 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>

**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.
### 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*.

<table>
<thead>
<tr>
<th>Tool Description</th>
<th>Location</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified CM Assistant server trace files</td>
<td>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:</td>
</tr>
<tr>
<td></td>
<td>• Use the CLI command <code>file get activelog tomcat/logs/ipma/log4j</code>.</td>
</tr>
<tr>
<td></td>
<td>• Use the trace collection features in the Cisco Unified Real-Time Monitoring Tool (RTMT). For more information, see the <em>Cisco Unified Real-Time Monitoring Tool Administration Guide</em>.</td>
</tr>
<tr>
<td></td>
<td>You can enable debug tracing by choosing <code>Cisco Unified Serviceability &gt; Trace &gt; Configuration</code>.</td>
</tr>
<tr>
<td>Cisco IPMA client trace files</td>
<td>$INSTALL_DIR/logs/ACLog*.txt on the client desktop, in the same location where the Unified CM Assistant assistant console resides.</td>
</tr>
<tr>
<td></td>
<td>To enable debug tracing, go to the <code>Settings</code> dialog box in the Assistant Console. In the <code>Advanced</code> panel, check the <code>Enable Trace</code> check box.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> This check box enables only debug tracing. Error tracing always remains On.</td>
</tr>
<tr>
<td>Cisco IPMA client install trace files</td>
<td>$INSTALL_DIR/InstallLog.txt on the client desktop, in the same location where the Assistant Console resides.</td>
</tr>
<tr>
<td>Cisco IPMA Client AutoUpdater trace files</td>
<td>$INSTALL_DIR/UpdatedLog.txt on the client desktop, in the same location where the Unified CM Assistant Console resides.</td>
</tr>
<tr>
<td>Install directory</td>
<td>By default—<code>C:\Program Files\Cisco\Unified Communications Manager Assistant Console\</code></td>
</tr>
</tbody>
</table>
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 Cisco Unified Communications Manager Assistant route point was not configured properly.

**Solution 2**
Use wildcards to match the directory number of the Cisco Unified Communications Manager Assistant CTI route point and the primary directory numbers of all managers that are configured for Cisco Unified Communications Manager Assistant.

**Possible Cause 3**
The status window on the manager phone displays the message Filtering Down. This message can indicate that Cisco 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 Cisco Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

Possible Cause 2
The server address for the primary and secondary Cisco 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 Cisco 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 Cisco Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

Possible Cause 4
The Cisco 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 Cisco 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 Cisco 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 CTI Manager. 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 Cisco 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.
Solution

Ensure that the Cisco IP Manager Assistant service has been activated by checking the activation status of the service from Cisco Unified Serviceability > Tools > Service Activation.

If the Cisco IP Manager Assistant service has already been activated, restart the Cisco Unified Communications Manager Assistant from Cisco Unified Serviceability > Tools > Control Center—Feature Services.

Manager Is Logged Out While the Service Is Still Running

Problem

Although the manager is logged out of Cisco 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 Cisco 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 Cisco Unified Communications Manager. The system does not upgrade the Assistant Console automatically when you upgrade the Cisco 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 Cisco 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 Cisco Unified Communications Manager Assistant CTI routepoint does not have Call Forward No Answer enabled.

Solution

Perform the following procedure to properly configure the Cisco 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 Cisco 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 Cisco Unified Communications Manager user through Cisco Unified CM Administration.

You must run the user as an assistant or a manager by associating the Cisco Unified Communications Manager Assistant user information, which you access through Cisco Unified CM Administration under User Management > End User.
Voice Messaging Features

- Audible Message Waiting Indicator, page 165
- Immediate Divert, page 169
Audible Message Waiting Indicator

- Audible Message Waiting Indicator Overview, page 165
- Audible Message Waiting Indicator Prerequisites, page 165
- Audible Message Waiting Indicator Configuration Task Flow, page 165
- Audible Message Waiting Indicator Troubleshooting, page 167

Audible Message Waiting Indicator Overview

You can configure Audible Message Waiting Indicator (AMWI) to play a stutter dial tone on the Cisco Unified IP Phones 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 Phones that are running phone firmware Release 8.3(1) or later.

Audible Message Waiting Indicator 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>Generate a Phone Feature List, on page 7</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Audible Message Waiting Indicator Service Parameters, on page 166</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.

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>System</strong> &gt; <strong>Service Parameters</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>From the <strong>Server</strong> drop-down list, 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, choose <strong>Cisco CallManager</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>In the <strong>Clusterwide Parameters (Feature - General)</strong> section, choose the <strong>Audible Message Waiting Indication Policy</strong> service parameter. This parameter determines whether the Audible Message Waiting Indicator is turned on or off for all the devices in the cluster.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
<td></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</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</strong> &gt; <strong>Phone</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>In the <strong>Association</strong> section, click <strong>Add a new DN</strong>. The <strong>Directory Number Configuration</strong> window appears.</td>
<td></td>
</tr>
</tbody>
</table>
| Step 3 | Select the **Audible Message Waiting Indicator Policy**. Choose one of the following options:  
• **Off**  
• **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. | |
Configure Audible Message Waiting Indicator for a SIP Profile

Follow these steps to configure AMWI for a SIP profile.

**Procedure**

1. From Cisco Unified CM Administration, choose Device > Device Settings > SIP Profile. The Find and List SIP Profiles window appears.
2. Enter the search criteria to use and click Find. The window displays a list of SIP profiles that match the search criteria.
3. Click the SIP profile that you want to update. The SIP Profile Configuration window appears.
4. Check the Stutter Message Waiting check box to activate stutter dial tone when the phone is off hook and a message is waiting.
5. Click Save.
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.

**Solution**

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 Cisco 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 Cisco 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. See the System Configuration Guide for Cisco Unified Communications Manager for information on how to configure voicemail profiles and hunt pilots.

- 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></td>
<td>iDivert</td>
</tr>
<tr>
<td>Cisco Unified IP Phone 7900 Series</td>
<td></td>
<td>X</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

### 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 Immediate Divert Service Parameters, on page 171</td>
<td>Configure the service parameters to enable Immediate Divert across various devices and applications.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure a Softkey Template for Immediate Divert, on page 173</td>
<td>Create and configure a softkey template and add the iDivert softkey to that template.</td>
</tr>
</tbody>
</table>
| Step 3 | To Associate a Softkey Template with a Common Device Configuration, on page 174, complete the following subtasks:  
- Add a Softkey Template to the Common Device Configuration, on page 175  
- Associate a Common Device Configuration with a Phone, on page 175 | 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 4 | Associate a Softkey Template with a Phone, on page 176 | 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 fields in the **Service Parameter Configuration** window. See the Related Topics section for more information about the fields and their configuration options.
- **Step 5** Click **Save**.
## 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>
<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.</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></td>
<td>The default value for this service parameter is <strong>True</strong>.</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.</td>
</tr>
<tr>
<td></td>
<td>The default value for this service parameter is <strong>False</strong>.</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>
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

**Procedure**

**Step 1**

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

**Step 2**

Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.

d) Click **Save**.

**Step 3**

Perform this step to add softkeys to an existing template.

a) Enter search criteria and click **Find**.

b) Choose an existing template.

The **Softkey Template Configuration** window appears.

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

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. See the What to Do Next section for more information.

**What to Do Next**

Perform one of the following procedures:

- **Add a Softkey Template to the Common Device Configuration**, on page 175
- **Associate a Softkey Template with a Phone**, on page 176

**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 176

**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 175</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Common Device Configuration with a Phone, on page 175</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. The Find and List Common Device Configuration window appears.
Step 2 Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
   c) Click Save.
Step 3 Perform this step to add the softkey template to an existing Common Device Configuration.
   a) Enter search criteria and click Find.
   b) Choose an existing Common Device Configuration.
   The Common Device Configuration window appears.
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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.
   • If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.

What to Do Next
Associate a Common Device Configuration with a Phone, on page 175

Associate a Common Device Configuration with a Phone

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.
Step 2 Find the phone to which 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

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.

Procedure

**Step 1** From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.

**Step 2** Choose the phone to which you want to add the softkey template. The Phone Configuration window appears.

**Step 3** From the Softkey Template drop-down list, choose the template that contains the new softkey.

**Step 4** Click Save.

Interactions and Restrictions

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>
</tbody>
</table>
### Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Detail Records (CDR)</td>
<td>Immediate Divert uses the immediate divert code number in the <strong>Onbehalf</strong> of fields (for example, joinOnbehalfOf and lastRedirectRedirectOnBehalfOf) in CDR.</td>
</tr>
<tr>
<td>Call Park and Directed Call Park</td>
<td>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.</td>
</tr>
<tr>
<td>Conference</td>
<td>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, cBarge, and Join.</td>
</tr>
<tr>
<td>Hunt List</td>
<td>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.</td>
</tr>
<tr>
<td>Auto Call Pickup</td>
<td>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 True, 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.</td>
</tr>
</tbody>
</table>
# 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 Voicemail 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>
</tbody>
</table>
| Busy Voicemail System              | The iDivert detects a busy condition on the voicemail ports, when iDivert reaches a voicemail system over a local or SCCP connection.  
**Note** Immediate Divert cannot divert a call to a busy voicemail port; voicemail ports can exist as members of a route or hunt list.  
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.  
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 **Allow QSIG During iDivert clusterwide** service parameter is set to **True**, or the **Use Legacy Immediate Divert clusterwide** service parameter is set to **False**, Immediate Divert supports access to voicemail systems that can be reached over QSIG or SIP trunks.  
When the **Allow QSIG During iDivert clusterwide** service parameter is set to **False**, and the **Use Legacy Immediate Divert clusterwide** service parameter is set to **True**, Immediate Divert does not support access to voicemail systems over QSIG or SIP trunks.                                                                 |
| Malicious Caller ID                | System does not support using Malicious Caller ID and Immediate Divert features together.                                                                                                                |
Immediate Divert Troubleshooting

**Key is not active**

**Problem**
The phone displays this message when the user presses iDivert:

*Key is not active*

**Possible Cause**
The voice-messaging profile of the user who pressed iDivert does not have a voice-messaging pilot.

**Solution**
Configure a voice-messaging pilot in the user voice-messaging profile.
Temporary Failure

**Problem**
The phone displays this message when the user presses iDivert:
Temporary Failure

**Possible Cause**
The voice-messaging system does not work, or a network problem exists.

**Solution**
Troubleshoot your voice-messaging system. See troubleshooting or voice-messaging documentation.

Busy

**Problem**
The phone displays this message when the user presses iDivert:
Busy

**Possible Cause**
This message means that the voice-messaging system is busy.

**Solution**
Configure more voice-messaging ports or try again.
PART VII

Conferencing Features

- Ad Hoc Conferencing, page 183
- Meet-Me Conferencing, page 195
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>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add the Conference List, Join, and Remove Last Conference Party softkeys to a softkey template.</td>
</tr>
<tr>
<td><strong>Step 2</strong></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</td>
</tr>
</tbody>
</table>
### Purpose

Command or Action | Purpose
--- | ---
• Add a Softkey Template to a Common Device Configuration, on page 186  
• Associate a Common Device Configuration with a Phone, on page 187 | 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 187**

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 4

**Configure Ad Hoc Conferencing, on page 188**

Enable advanced conferencing, specify the maximum number of participants, and specify when to drop a conference connection.

### Step 5

**Configure Join Across Lines, on page 190**

Enable Join Across Lines to create a conference.

### Configure a 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 (ConList)</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

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; Softkey Template. The Softkey Template Configuration window appears.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Perform this step to create a new softkey template; otherwise, proceed to the next step.</td>
</tr>
<tr>
<td>a)</td>
<td>Click Add New.</td>
</tr>
<tr>
<td>b)</td>
<td>Select a default template and click Copy.</td>
</tr>
<tr>
<td>c)</td>
<td>In the Softkey Template Name field, enter a new name for the template.</td>
</tr>
<tr>
<td>d)</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Perform this step to add softkeys to an existing template.</td>
</tr>
<tr>
<td>a)</td>
<td>Enter search criteria and click Find.</td>
</tr>
<tr>
<td>b)</td>
<td>Choose an existing template. The Softkey Template Configuration window appears.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Check the Default Softkey Template check box to designate this softkey template as the default softkey template.</td>
</tr>
<tr>
<td>Note</td>
<td>If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Choose Configure Softkey Layout from the Related Links drop-down list in the upper right corner and click Go.</td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Select a Call State to Configure drop-down list, choose the call state for which you want the softkey to display.</td>
</tr>
<tr>
<td>Step 7</td>
<td>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.</td>
</tr>
<tr>
<td>Step 8</td>
<td>To display the softkey in additional call states, repeat the previous step.</td>
</tr>
<tr>
<td>Step 9</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 10</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>
<tr>
<td></td>
<td>• If you created a new softkey template, associate the template with the devices and then restart them. See the What to Do Next section for more information.</td>
</tr>
</tbody>
</table>

What to Do Next

Perform one the following procedures:

• Add a Softkey Template to a Common Device Configuration, on page 186
• Associate a Softkey Template with a Phone, on page 187
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 187.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Add a Softkey Template to a Common Device Configuration, on page 186</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate a Common Device Configuration with a Phone, on page 187</td>
<td></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. The Find and List Common Device Configuration window appears.

**Step 2** Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
   c) Click Save.

**Step 3** Perform this step to add the softkey template to an existing Common Device Configuration.
   a) Enter search criteria and click Find.
   b) Choose an existing Common Device Configuration.
   The Common Device Configuration window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.
• If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.

---

**What to Do Next**

Associate a Common Device Configuration with a Phone, on page 187

---

**Associate a Common Device Configuration with a Phone**

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**  
Find the phone to which 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.

**Procedure**

**Step 1**  
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**  
Choose the phone to which you want to add the softkey template. The **Phone Configuration** window appears.

**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 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

Step 1 From Cisco Unified CM Administration, choose System > Service Parameters.
Step 2 From the Server drop-down list, choose the server.
Step 3 From the Service drop-down list, choose Cisco CallManager.
Step 4 Configure the fields in the Clusterwide Parameters (Features - Conference) area. See the Related Topics section for more information about the fields and their configuration options.
Step 5 Click Save.

Related Topics

Ad Hoc Conferencing Service Parameters, on page 189
### Ad Hoc Conferencing Task Flow

#### Ad Hoc Conferencing Service Parameters

<table>
<thead>
<tr>
<th>Service Parameters</th>
<th>Description</th>
</tr>
</thead>
</table>
| Drop Ad Hoc Conference           | 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.  

  - **Never**—The conference does not get dropped. (Cisco recommends that you use the default option to avoid unintentional termination of a conference).

  - **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. Cisco Unified Communications Manager releases all resources that are assigned to the conference.

  - **When Conference Controller Leaves**—The active conference terminates when the primary controller (conference creator) hangs up. Cisco Unified Communications Manager releases all resources that are assigned to the conference.

  **Note**  
  Cisco recommends that you set this service parameter to **Never**. Any other setting can result in unintentional termination of a conference.

  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.

| Maximum Ad Hoc Conference        | This parameter specifies the maximum number of participants that are allowed in a single Ad Hoc conference.  

  **Default Value:** 4

<p>| Advanced Ad Hoc Conference Enabled | 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. |</p>
<table>
<thead>
<tr>
<th>Service Parameters</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<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>
<tr>
<td>Enable Click-to-Conference for Third-Party Applications</td>
<td>This parameter determines whether the Click-to-Conference functionality over the SIP trunk is enabled on Cisco 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. Enabling this parameter could negatively impact CTI applications that are not coded to support this feature. Default value: False</td>
</tr>
</tbody>
</table>

**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 7

**Procedure**

1. From the Cisco Unified CM Administration, choose **Device > Device Settings > Default Device Profile**. The **Default Device Profile Configuration** window is displayed.
2. From the **Device Profile Type** drop-down list, choose the phone model.
3. From the **Device Protocol** drop-down list, choose the relevant SCCP or SIP protocol.
4. Set the **Join Across Lines** to On.
5. Click **Save**.
## Conference Interactions and Restrictions

### Conference Interactions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference by Using <strong>cBarge</strong></td>
<td>Initiate a conference by pressing the <strong>cBarge</strong> 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 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:
### Conference Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ad Hoc conference</td>
<td>Cisco Unified Communications Manager supports a maximum of 100 simultaneous Ad Hoc conferences for each Cisco Unified Communications Manager server.</td>
</tr>
<tr>
<td></td>
<td>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>

**Ad Hoc conference on SIP phones:**

- Cisco Unified IP Phone 7911
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7961
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971

<table>
<thead>
<tr>
<th>Restrictions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified Communications Manager uses &quot;beep&quot; and &quot;beep beep&quot; 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 &quot;beep beep&quot;. Users might not hear the beeps because of the time it takes Cisco 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 Phones 7911, 7941, 7961, 7970, and 7971.</td>
</tr>
<tr>
<td>Feature</td>
</tr>
<tr>
<td>---------------------------------------------</td>
</tr>
<tr>
<td>Ad Hoc conference on SIP phones:</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>
Meet-Me Conferencing

Meet-Me Conferencing Overview

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

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add the Meet-Me softkey to a softkey template.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Optional. To make the softkey template available to phones, you must complete either this step or the following step.</td>
</tr>
<tr>
<td>Step 3</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</td>
</tr>
</tbody>
</table>
Configure a Softkey Template for Meet-Me Conferencing

Use this procedure to make the Meet Me software key available in the off hook call state.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; Softkey Template. The Softkey Template Configuration window appears.</th>
</tr>
</thead>
</table>
| Step 2 | Perform this step 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) In the Softkey Template Name field, enter a new name for the template.  
   d) Click Save. |
| Step 3 | Perform this step to add softkeys to an existing template.  
   a) Enter search criteria and click Find.  
   b) Choose an existing template. The Softkey Template Configuration window appears. |
| 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 | To display the softkey in additional call states, repeat the previous step. |
| 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. See the What to Do Next section for more information.

What to Do Next
Perform one the following procedures:
• Add a Softkey Template to a Common Device Configuration, on page 197
• Associate a Softkey Template with a Phone, on page 199

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 199.

Before You Begin
Configure a Softkey Template for Meet-Me Conferencing, on page 196

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 197</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Common Device Configuration with a Phone, on page 198</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. The Find and List Common Device Configuration window appears.

Step 2
Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
c) Click Save.

Step 3 Perform this step to add the softkey template to an existing Common Device Configuration.
a) Enter search criteria and click Find.
b) Choose an existing Common Device Configuration.
The Common Device Configuration window appears.

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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.

• If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.

What to Do Next

Associate a Common Device Configuration with a Phone, on page 198

Associate a Common Device Configuration with a Phone

Before You Begin

Add a Softkey Template to a Common Device Configuration, on page 197

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.

Step 2 Find the phone to which 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 199
**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 Meet-Me Conferencing, on page 196

**Procedure**

1. **Step 1**
   From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.

2. **Step 2**
   Choose the phone to which you want to add the softkey template. The Phone Configuration window appears.

3. **Step 3**
   From the Softkey Template drop-down list, choose the template that contains the new softkey.

4. **Step 4**
   Click Save.

5. **Step 5**
   Press Reset to update the phone settings.

**What to Do Next**

Configure a Meet-Me Conferencing Number, on page 199

**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**

1. **Step 1**
   From Cisco Unified CM Administration, choose Call Routing > Meet-Me Number/Pattern. The Find and List Meet-Me Numbers window appears.

2. **Step 2**
   Enter the appropriate search criteria and click Find. All matching records are displayed.

3. **Step 3**
   In the list of records, click the link for the record that you want to view.

4. **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**.

---

**Related Topics**

[Meet-Me Number and Pattern Settings, on page 200](#)

---

**Meet-Me Number and Pattern Settings**

The following table describes the Meet-Me number and pattern settings.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| **Directory Number or Pattern** | 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. |
| **Description**               | The description can include up to 50 characters in any language, but it cannot include double quotation marks ("), percentage sign (%), ampersand (&), or angle brackets (<>). |
| **Partition**                 | 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 <None> 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.  
**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 Cisco Unified Communications Manager cluster. |
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Minimum Security</td>
<td>Choose the minimum Meet-Me conference security level for this Meet-Me number or pattern from the drop-down list box.</td>
</tr>
<tr>
<td>Level</td>
<td>• Choose <strong>Authenticated</strong> to block participants with nonsecure phones from joining the conference.</td>
</tr>
<tr>
<td></td>
<td>• Choose <strong>Encrypted</strong> to block participants with authenticated or nonsecure phones from joining the conference.</td>
</tr>
<tr>
<td></td>
<td>• Choose <strong>Non Secure</strong> to allow all participants to join the conference.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>To invoke this feature, ensure that you have a secure conference bridge that is configured and available. For more information, see the <em>Cisco Unified Communications Manager Security Guide</em>.</td>
</tr>
</tbody>
</table>

**Meet-Me Conferencing Restrictions**

Cisco Unified Communications Manager supports a maximum of 100 simultaneous Meet-Me conferences for each Cisco 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.
PART VIII

Placing Calls

- Call Back, page 205
- Hotline, page 217
- Speed Dial and Abbreviated Dial, page 231
- WebDialer, page 235
- Paging, page 251
- Intercom, page 281
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 Cisco 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 Cisco 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 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 6: Cisco Unified IP Phones That Use CallBack Softkeys and Buttons

<table>
<thead>
<tr>
<th>Cisco Unified IP 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 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>

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configure Softkey Template for CallBack</td>
<td>Perform this step to add CallBack softkey to template and configure the softkey using the Common Device Configuration or phone.</td>
<td></td>
</tr>
</tbody>
</table>
Perform this step to add and configure the CallBack button to a phone.

### 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</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>From Cisco Unified CM Administration, choose Device &gt; Device Settings &gt; Softkey Template. The Softkey Template Configuration window appears.</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Perform this step to create a new softkey template; otherwise, proceed to the next step.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>a) Click Add New.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>b) Select a default template and click Copy.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>c) In the Softkey Template Name field, enter a new name for the template.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>d) Click Save.</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Perform this step to add softkeys to an existing template.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>a) Enter search criteria and click Find.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>b) Choose an existing template.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>The Softkey Template Configuration window appears.</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>Check the Default Softkey Template check box to designate this softkey template as the default softkey template.</td>
<td></td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.</td>
<td></td>
</tr>
</tbody>
</table>
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 To display the softkey in additional call states, repeat the previous step.

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. See the **What to Do Next** section for more information.

**What to Do Next**

Perform one of the following procedures:

- Associate CallBack Softkey Template with a Common Device Configuration, on page 208
- Associate CallBack Softkey Template with Phone, on page 210

**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 210.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Add CallBack Softkey Template to the Common Device Configuration, on page 209</td>
<td>Perform this step to add CallBack softkey template to the Common Device Configuration.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate a Common Device Configuration with a Phone, on page 209</td>
<td>Perform this step to link the CallBack softkey Common Device Configuration to a phone.</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**. The **Find and List Common Device Configuration** window appears.

**Step 2**
Perform this step 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) In the **Name** field, enter a name for the Common Device Configuration.
c) Click **Save**.

**Step 3**
Perform this step to add the softkey template to an existing Common Device Configuration.

a) Enter search criteria and click **Find**.
b) Choose an existing Common Device Configuration.

The **Common Device Configuration** window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.
- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.

What to Do Next

Associate a Common Device Configuration with a Phone, on page 209

Associate a Common Device Configuration with a Phone

Procedure

**Step 1**
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**
Find the phone to which 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**

<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>The <strong>Find and List Phones</strong> window appears.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Choose the phone to which you want to add the softkey template.</td>
<td>The <strong>Phone Configuration</strong> window appears.</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>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Save</strong>.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Press <strong>Reset</strong> to update the phone settings.</td>
<td></td>
</tr>
</tbody>
</table>

**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 a Phone Button Template for Call Back, on page 210</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 211</td>
<td>Perform this step to configure the CallBack button for a phone.</td>
</tr>
</tbody>
</table>

**Configure a Phone Button Template for Call Back**

Optional. Follow this procedure when you want to assign features to line or speed dial keys.

**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; Device Settings &gt; Phone Button Template</strong>.</td>
<td>The <strong>Find and List Phone Button Templates</strong> window appears.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong>.</td>
<td>The window displays a list of templates for the supported phones.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Perform this step if you want to create a new phone button template; otherwise, proceed to the next step.</td>
<td></td>
</tr>
</tbody>
</table>
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 this step if you want to add phone buttons to an existing template.
a) Enter search criteria and click Find.
b) Choose an existing template.
The Phone Button Template Configuration window appears.

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. See the What to Do Next section for more information.

What to Do Next

Associate a Button Template with a Phone, on page 211

Associate a Button Template with a Phone

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window is displayed.

Step 2 From the Find and List Phones window, click Find.
A list of phones that are configured on the Cisco Unified Communications Manager is displayed.

Step 3 Choose the phone to which you want to add the phone button template.
The Phone Configuration window appears.

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 and Restrictions

## 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>
</tbody>
</table>
| CallBack notification with phones running SIP | 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 Cisco 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 Cisco Unified Communications Manager receives from the phone. After the phone sends the SIP INVITE to Cisco Unified Communications Manager and the phone goes on-hook, Cisco Unified Communications Manager sends an audio and CallBack notification screen to the Cisco Unified IP Phone 7960 and 7940 (SIP) user. |
| Do Not Disturb (DND)            | CallBack would work normally in case or when **DND-Reject** is set to **Off** at the originating or the terminating end. The behavior differs only when **DND-Reject** is set to **On**.  
- **DND-Reject On on Originating end**—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.  
- **DND-Reject On on Terminating end**—User A calls User B, and User B has set **DND-Reject** to **On**. 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 **DND-Reject** to **Off**. |
| Cisco Extension Mobility        | 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. |
Call Back Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<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**
Cisco 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 Cisco 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.
CHAPTER 20

Hotline

This chapter describes how to use and configure the Hotline feature.

- Hotline Overview, page 217
- System Requirements for Hotline, page 218
- Hotline Configuration Task Flow, page 218
- Hotline Troubleshooting, page 229

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 Cisco Unified Communications Manager:

- Cisco 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://www.cisco.com/go/cfn.
You do not need a Cisco.com account to access Cisco Feature Navigator.

Hotline 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 7</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> Create Custom Softkey Template, on page 219</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> Configure Hotline on Phones, on page 219</td>
<td>Enable the phone as a Hotline device.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure Route Class Signaling Task Flow, on page 220</td>
<td>Configure route class signaling to support the Hotline feature.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure Hotline to Call Only or Receive Only Task Flow, on page 224</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> Configure Call Screening with a Calling Search Space, on page 226</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.

Cisco Unified Communications Manager includes standard softkey templates for call processing and applications. When creating custom softkey templates, copy the standard templates and make modifications as required.

**Procedure**

**Step 1** Choose Device > Device Settings > Softkey Template.

**Step 2** Click Add New.

**Step 3** From the drop-down list, select a softkey template and click Copy to create a new template.

**Step 4** In the Softkey Template Name field, enter a unique name to identify the softkey template.

**Step 5** 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 ("), percentage sign (%), ampersand (&), backslash (\), or angle brackets (<>).

**Step 6** To designate this softkey template as the standard softkey template, check the Default Softkey Template check box.

**Note** 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.

**Step 7** Click Save.

The softkey template gets copied, and the Softkey Template Configuration window redisplay.

**Step 8** (Optional) Click the Add Application button.

**Step 9** Configure the positions of the softkeys on the Cisco Unified IP Phone LCD screen.

**Step 10** To save your configuration, click Save.

**What to Do Next**

Configure Hotline on Phones, on page 219

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 219.
Procedure

Step 1  In Cisco Unified CM Administration, choose Device > Phone.
Step 2  Click Find and select the phone that you want to enable as a Hotline device.
Step 3  Check the Hotline Device check box.
Step 4  If you have created a custom softkey template specifically for the Hotline phone, from the Softkey Template drop-down list, choose the softkey template.
Step 5  Click Save.

Note  You can also assign a softkey template to a Device Pool and then assign that Device Pool to the phone.

What to Do Next
Configure Route Class Signaling Task Flow, on page 220

Configure Route Class Signaling Task Flow
Perform this task flow to configure route class signaling for Hotline calls.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Enable Route Class Signaling in the Cluster, on page 221</td>
<td>Set the route class signaling clusterwide defaults for trunks and gateways to enabled.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> 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><strong>Step 2</strong> Enable Route Class Signaling on Trunks, on page 221</td>
<td>Enable route class signaling on an individual trunk.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Enable Route Class Signaling on Gateways, on page 222</td>
<td>Enable route class signaling on an MGCP T1/CAS or MGCP PRI gateway.</td>
</tr>
<tr>
<td><strong>Step 4</strong> Configure Signaling Labels for the Hotline Route Class, on page 222</td>
<td>Configure SIP signaling labels for Hotline route classes.</td>
</tr>
<tr>
<td><strong>Step 5</strong> Configure the Route Class on Hotline Route Patterns, on page 223</td>
<td>Configure the route class on the route patterns that are routing your Hotline calls.</td>
</tr>
<tr>
<td><strong>Step 6</strong> Configure the Route Class on Hotline Translation Patterns, on page 224</td>
<td>Optional. 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.

Note

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
In 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 221
Enable Route Class Signaling on Gateways, on page 222

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 221 procedure to use a clusterwide service parameter to configure the default route class signaling settings for all trunks in the cluster.

Procedure

Step 1
In 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.
What to Do Next

Enable Route Class Signaling on Gateways, on page 222

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 221 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 221 procedure to configure route class signaling for individual trunks.

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Gateways.
Step 2 Click Find and select the gateway on which you want to configure route class signaling.
Step 3 From the Route Class Signaling Enabled drop-down list box, choose one of the following options:
  • Default—This gateway uses the setting from the clusterwide Route Class Signaling Enabled service parameter.
  • Off—Route class signaling is disabled on this gateway.
  • On—Route class signaling is enabled on this gateway.
Step 4 If you want to encode voice route class for voice calls, check the Encode Voice Route Class check box.
Step 5 Click Save.

What to Do Next

Configure Signaling Labels for the Hotline Route Class, on page 222

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 221.
### Procedure

**Step 1**  
In 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**  
Click **Advanced**.

**Step 5**  
In the **SIP Route Class Naming Authority** 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 **cisco.com**.

**Step 6**  
In the **SIP Hotline Voice Route Class Label** service parameter field, enter a label to represent the Hotline Voice route class. The default value is **hotline**.

**Step 7**  
In the **SIP Hotline Data Route Class Label** service parameter field, enter a label to represent the Hotline Data route class. The default value is **ccdata**.

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

### What to Do Next

**Configure the Route Class on Hotline Route Patterns**, on page 223

### Configure the Route Class on Hotline Route Patterns

This procedure describes call routing instructions that are specific to Hotline devices. For detailed information on how to configure route patterns and translation patterns in your network, see the *Cisco Unified Communications Manager System Configuration Guide*.

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 222

Before you perform this procedure, it is expected that your network call routing is set up with route patterns.

### Procedure

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

**Step 2**  
Click **Find** to display a list of route patterns in your network.

**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**.

### What to Do Next

**Configure the Route Class on Hotline Translation Patterns**, on page 224
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 223 procedure.

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 Call Routing &gt; Translation Pattern.</td>
<td>Create two partitions: one should be empty and the other will be assigned to a new CSS.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Find to display the translation patterns in your cluster.</td>
<td></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>Step 1</td>
<td>Configure Partitions for Hotline Call Only Receive Only, on page 224</td>
<td>Create two partitions: one should be empty and the other will be assigned to a new CSS.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Calling Search Space for Hotline Call Only Receive Only, on page 225</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 226  
  • Configure Receive Only on Hotline Phone, on page 226 | 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**

**Step 1**
In Cisco Unified CM Administration, choose **Call Routing > Class of Control > Partitions**.

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

**Step 3**
Create a new partition.

**Step 4**
Enter a unique name and description for the partition. For example, **IsolatedPartition**.

*Note*  This partition will not be assigned to any CSS.

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

**Step 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.

---

**What to Do Next**

Configure Calling Search Space for Hotline Call Only Receive Only, on page 225

**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 224

**Procedure**

**Step 1**
In 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 **EmptyPartition** 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 226
- Configure Receive Only on Hotline Phone, on page 226
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 225

**Procedure**

**Step 1** In 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.

**Before You Begin**

Configure Calling Search Space for Hotline Call Only Receive Only, on page 225

**Procedure**

**Step 1** In 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 CSS that you created in the previous procedure.
**Step 4** Click Save.

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.

**Note** 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.
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

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Configure Partitions for Hotline Call Screening, on page 227</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Create Calling Search Space for Hotline Call Screening, on page 228</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</td>
<td>Configure Hotline Phones for Call Screening, on page 229</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 In Cisco Unified Communications Manager 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 Related Topics section 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 is available to receive an incoming call.

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

---

**What to Do Next**

Create Calling Search Space for Hotline Call Screening, on page 228

---

### 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 227

### Procedure

**Step 1**  From Cisco Unified CM Administration, select **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 Hotline Phones for Call Screening, on page 229
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 228

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 Hotline phone.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the Calling Search Space drop-down list box, select the new calling search space that you created for the Hotline call screening list.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save.</td>
</tr>
<tr>
<td>Step 5</td>
<td>From the left navigation pane, click the phone line that you want to use for Hotline calls. The Directory Number Configuration window displays.</td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Route Partition drop-down list box, select a partition that is included in the calling search space that you set up.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Hotline Troubleshooting

The following table provides troubleshooting information for cases where hotline calls do not dial correctly.

Table 7: Troubleshooting 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 8: 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.
CHAPTER 21

Speed Dial and Abbreviated Dial

- Speed Dial and Abbreviated Dial Overview, page 231
- Speed Dial and Abbreviated Dial Configuration Task Flow, page 232

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.

Pause In Speed Dial

You can use Speed Dial to reach destinations that require a Forced Authorization Code (FAC), Client Matter Code (CMC), dialing pauses, or additional digits (such as a user extension, a meeting access code, or a voice mail password). When you press 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.

To include dialing pauses in the speed dial, include a comma (,) as part of the speed-dial string. A comma acts as a pause duration for post connect DTMF digits and also as a delimiter between destination address digits and FAC, CMC codes. Each comma you include represents an additional pause of 2 seconds. For example, two commas (,,) represent a pause of 4 seconds. It also allows you to separate FAC and CMC from the other digits in the speed-dial string.
### 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 7</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 232</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.

#### Procedure

**Step 1**

In Cisco Unified Communications Manager Administration, choose Device > Phone. Enter your search criteria and click Find. Choose the phone for which you want to configure speed dial buttons. The Phone Configuration window appears.

**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.
WebDialer

WebDialer Overview

Cisco WebDialer is installed on a Cisco Unified Communications Manager node and used along with Cisco 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.

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/ucmuser

WebDialer Prerequisites

Cisco WebDialer requires the following software components:

• CTI-supported Cisco Unified IP Phones
## WebDialer Configuration Task Flow

### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Activate WebDialer, on page 237</td>
<td>Activate the WebDialer service.</td>
</tr>
<tr>
<td>2</td>
<td>(Optional) Enable WebDialer Tracing, on page 237</td>
<td>To view WebDialer traces, enable tracing.</td>
</tr>
<tr>
<td>3</td>
<td>(Optional) Configure WebDialer Servlet, on page 238</td>
<td>Configure the WebDialer servlet.</td>
</tr>
<tr>
<td>4</td>
<td>(Optional) Configure Redirector Servlet, on page 238</td>
<td>If you have multi cluster applications that you develop using HTML over HTTPS interfaces, configure the Redirector servlet.</td>
</tr>
<tr>
<td>5</td>
<td>(Optional) Configure WebDialer Application Server, on page 239</td>
<td>To configure Redirector for Cisco WebDialer.</td>
</tr>
</tbody>
</table>
| 6    | (Optional) To Configure Secure TLS Connection to CTI, on page 239, complete the following sub tasks:  
  - Configure WDSecureSysUser Application User, on page 240  
  - (Optional) Configure CAPF Profile, on page 140  
  - (Optional) Configure Cisco IP Manager Assistant, on page 142 | 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. |
| 7    | Configure Language Locale for WebDialer, on page 242 | Determine which language WebDialer displays by setting the locale field in the Cisco Unified Communications Self Care Portal menu. |
| 8    | Configure WebDialer Alarms, on page 243 | If there are any issues with the Web Dialer feature it alerts the administrator. |
| 9    | (Optional) Configure Application Dial Rules, on page 243 | If your application requires multiple clusters, configure application dial rules. |
| 10   | Add Users to Standard CCM End User Group, on page 243 | Add each WebDialer user to the Standard End User Group for Cisco Unified Communications Manager. |
| 11   | (Optional) To Configure Proxy User, on page 244, complete the following sub tasks:  
  - Add a WebDialer End User, on page 245  
  - Assign Authentication Proxy Rights, on page 245 | If you use makeCallProxy HTML over HTTP interface to develop an application for using Cisco WebDialer, create a proxy user. |
**Activate WebDialer**

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From the navigation area of the Cisco Unified Communications Manager application, choose <strong>Cisco Unified Serviceability</strong> and click <strong>Go</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Choose <strong>Tools &gt; Service Activation</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Servers</strong> drop-down list, choose the Cisco Unified Communications Manager server that is listed.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From <strong>CTI Services</strong>, check the <strong>Cisco WebDialer Web Service</strong> check box.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td>Step 6</td>
<td>From Cisco Unified Serviceability, choose <strong>Tools &gt; Control Center - Feature Services</strong> 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.</td>
</tr>
</tbody>
</table>

**What to Do Next**

Enable WebDialer Tracing, on page 237

**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*.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>From the navigation drop-down list of the Cisco Unified Communications Manager application, choose <strong>Cisco Unified Serviceability</strong> and then click <strong>Go</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Choose <strong>Trace &gt; Configuration</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>From the <strong>Server</strong> drop-down list, choose the server on which to enable tracing.</td>
</tr>
<tr>
<td>Step 4</td>
<td>From the <strong>Service</strong> drop-down list, choose the <strong>Cisco WebDialer Web Service</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the <strong>Trace Configuration</strong> window, change the trace settings according to your troubleshooting requirements.</td>
</tr>
</tbody>
</table>

**Note**
For more information about WebDialer trace configuration settings, see the *Cisco Unified Serviceability Administration Guide*.  

---

**WebDialer Configuration Task Flow**

---

Feature Configuration Guide for Cisco Unified Communications Manager, Release 10.5(2)
Step 6  Click Save.

What to Do Next
Configure WebDialer Servlet, on page 238

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.

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.

What to Do Next
Configure Redirector Servlet, on page 238

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.

Procedure

Step 1  From Cisco Unifies CM Administration, choose System > Service Parameters.
Step 2  From the Server drop-down list, choose the Cisco Unified Communications Manager server on which to configure the Redirector Servlet.
Step 3  From the Service drop-down list, choose the 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.
For more information on WebDialer Web Service, see the Cisco Unified Serviceability Administration Guide.
What to Do Next
Configure WebDialer Application Server, on page 239

Configure WebDialer Application Server

Application server is required to configure the Redirector Servlet. Redirector is required only when you have multiple Cisco Unified Communications Manager servers configured in a cluster.

Procedure

Step 1 In the Cisco Unified Communications Manager Administration Application server window, choose System > Application Server.
Step 2 From the Application Server Type drop-down list box, 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 the Cisco Unified Communications Manager Security Guide.
• 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 the Cisco Unified Communications Manager Security Guide.
• Activate the Cisco Certificate Authority Proxy Function service on the first node.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure WDSecureSysUser Application User, on page 240</td>
<td>Configure a WDSecureSysUser application user.</td>
</tr>
<tr>
<td>Step 2 Configure CAPF Profile, on page 140</td>
<td>Configure a CAPF profile for the WDSecureSysUser application user.</td>
</tr>
<tr>
<td>Step 3 Configure Cisco IP Manager Assistant, on page 142</td>
<td>Configure service parameters for the Cisco IP Manager Assistant service.</td>
</tr>
</tbody>
</table>
Configure WDSecureSysUser Application User

**Procedure**

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

**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.

**What to Do Next**

Configure CAPF Profile, on page 140

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**  
In Cisco Unified CM Administration, choose User Management > Application User CAPF Profile.

**Step 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.

**What to Do Next**

Configure Cisco IP Manager Assistant, on page 142

**Related Topics**

CAPF Profile Settings, on page 141
## 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 <em>End User CAPF Profile</em> 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 <em>Application User CAPF Profile</em> 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. |
| **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. |
### Configure Cisco IP Manager Assistant

**Procedure**

**Step 1** In 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.

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.

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

**Step 6** Repeat the procedure on each server on which the service is active.

### 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.

**Procedure**

**Step 1** In 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.

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**.

Configure Application Dial Rules

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.

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 (Optional) Add a WebDialer End User, on page 245</td>
<td>Add a new user. If the user exists, you can proceed to the next task.</td>
</tr>
<tr>
<td>Step 2 Assign Authentication Proxy Rights, on page 245</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 245

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>
</tbody>
</table>
| Client Matter Codes and Forced Authorization Codes | 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. |

WebDialer Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restrictions</th>
</tr>
</thead>
</table>
| Phones                   | Cisco WebDialer supports phones that run Skinny Client Control Protocol (SCCP) and Session Initiation Protocol (SIP) that Cisco Computer Telephony Integration (CTI) supports.  
  **Note** Few older phone models do not support Cisco WebDialer that run SIP. |
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.
Paging

• Paging Overview, page 251
• Paging Prerequisites, page 252
• Basic Paging Configuration Task Flow, page 252
• Advanced Notification Paging Task Flow, page 253

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 Advanced Notification
• InformaCast Mobile

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 a variety of devices and systems with text and audio messages.

Some of the features include:

• Ability to reach analog PA systems and IP speakers
• Support for Cisco Jabber
• 911 emergency call monitoring, recording, and notification
• Dynamic conference call
• Event accountability with message confirmation and reporting
• Implementation with panic buttons
• Building evacuation or lockdown
• Facilities integration (control lighting, door locks)
• Security integration (motion detectors, access, fire)
• General purpose and shift or bell scheduling

Users must 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.

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 multicasts.

For a list of Cisco Unified IP Phones that support paging, refer to the *[Cisco Unified IP Phones](http://www.singlewire.com/compatibility-matrix.html)* section of the Singlewire Compatibility Matrix at:


**Basic Paging Configuration Task Flow**

Perform the following tasks to configure Cisco Unified Communications Manager to integrate with Cisco Paging Server for an InformaCast Basic Paging deployment.
### Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
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<td>Enable SNMP Service, on page 255</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 258</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 259</td>
<td>Configure a device pool.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure Route Partition for InformaCast Paging, on page 260</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 261</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 265</td>
<td>Configure CTI ports.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Configure Access Control Group with AXL Access, on page 267</td>
<td>Configure an AXL user group/access control group.</td>
</tr>
<tr>
<td>Step 8</td>
<td>Configure Application User for Paging, on page 267</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:</td>
<td>You can enable web access on a individual phone or use a Common Phone Profile to enable web access for a group of phones that use that profile.</td>
</tr>
<tr>
<td></td>
<td>- Enable Web Access for a Phone, on page 268</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- Enable Web Access for Common Phone Profile, on page 269</td>
<td></td>
</tr>
<tr>
<td>Step 10</td>
<td>Configure Authentication URL, on page 269</td>
<td>Configure the Cisco Unified Communications Manager authentication URL to point to InformaCast so that when InformaCast pushes broadcasts to Cisco Unified IP Phones, the phones will authenticate with InformaCast.</td>
</tr>
</tbody>
</table>


---

**Advanced Notification Paging Task Flow**

Perform the following tasks to configure Cisco Unified Communications Manager to integrate with an InformaCast Advanced Notification paging deployment. An Advanced Notification deployment can include the following features:

- InformaCast paging
- CallAware
- PushToTalk
- Legacy paging devices
- Plugins such as Facebook, Twitter and conferencing

## Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
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<td>Configure SNMP for Paging, on page 255</td>
<td>Configure SNMP services in Cisco Unified Communications Manager.</td>
</tr>
<tr>
<td>2</td>
<td>Configure Media Resources for Legacy Paging, on page 257</td>
<td>Optional. If you want to connect to legacy paging devices, configure a media resource group that includes an MTP.</td>
</tr>
<tr>
<td>3</td>
<td>Configure Region for Paging, on page 258</td>
<td>Configure a region for paging with G.711 as the default codec and a device pool for paging devices.</td>
</tr>
<tr>
<td>4</td>
<td>Configure Partitions and Calling Search Spaces for Paging, on page 260</td>
<td>Configure partitions and calling search spaces for your paging deployment.</td>
</tr>
<tr>
<td>5</td>
<td>Configure CTI Route Points for CallAware, on page 262</td>
<td>If you are deploying CallAware, configure CTI route points for each CallAware redirect.</td>
</tr>
<tr>
<td>6</td>
<td>Enable the Built in Bridge using one of the following procedures:</td>
<td>If you are deploying CallAware, enable the Built in Bridge using a clusterwide service parameter, or on the phone itself.</td>
</tr>
<tr>
<td></td>
<td>- Enable Built in Bridge for Cluster, on page 263</td>
<td>Note: The settings in the Phone Configuration window for an individual phone override the clusterwide service parameter.</td>
</tr>
<tr>
<td></td>
<td>- Enable Built in Bridge for a Phone, on page 264</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>Configure CTI Ports for Paging, on page 265</td>
<td>Configure CTI ports for InformaCast paging and CallAware redirects.</td>
</tr>
<tr>
<td>8</td>
<td>Configure SIP Trunk for Legacy Paging Devices, on page 266</td>
<td>If you are connecting to legacy paging devices, configure a SIP trunk that connects to your legacy devices.</td>
</tr>
<tr>
<td>9</td>
<td>Configure Access Control Group with AXL Access, on page 267</td>
<td>Configure an access control group that includes access to the AXL API.</td>
</tr>
<tr>
<td>10</td>
<td>Configure Application User for Paging, on page 267</td>
<td>Configure an application user. You must configure different application users for InformaCast, CallAware, and PushToTalk.</td>
</tr>
<tr>
<td>11</td>
<td>Perform one of the following procedures:</td>
<td>Enable web access for your phones. You can enable web access on an individual phone, or use a Common Phone Profile to enable web access for a group of phones.</td>
</tr>
<tr>
<td></td>
<td>- Enable Web Access for a Phone, on page 268</td>
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</table>
Configure the Cisco Unified Communications Manager authentication URL to point to InformaCast so that when InformaCast pushes broadcasts to Cisco Unified IP Phones, the phones will authenticate with InformaCast.

Configure PushToTalk service integration for your Cisco Unified IP Phones.

Optional. Assign directory URIs to your paging directory numbers. This allows you to implement URI dialing in your paging deployment.

For detailed information on the InformaCast Advanced Notification deployment options, refer to your InformaCast Advanced Notification product documentation. For help, visit Singlewire's website at http://www.singlewire.com.

**Configure SNMP for Paging**

Perform the following tasks to configure SNMP services in the cluster for either Basic Paging or Advanced Notification deployments.

**Procedure**

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<td><strong>Step 2</strong></td>
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</table>

**Enable SNMP Service**

To configure paging in either Basic Paging or Advanced Notification deployments, 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 all nodes in the cluster.
- 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 and Cisco CallManager.
Step 4 For at least one server in the cluster, check the check boxes that correspond to the Cisco CTIManager and Cisco AXL Web Service services.
Step 5 Click Save.
Step 6 Click OK.
Step 7 Repeat the previous steps for all nodes in the cluster.

What to Do Next
Create an InformaCast SNMP Community String, on page 256

Create an InformaCast SNMP Community String
Perform this procedure for Basic Paging or Advanced Notification deployments to set up an SNMP community string.

Before You Begin
Enable SNMP Service, on page 255

Procedure

Step 1 In Cisco Unified Serviceability, choose SNMP > V1/V2c > Community String.
Step 2 From the Server drop-down list box, choose a server and click Find.
Step 3 Click Add New.
Step 4 In the Community String Name field, enter ICVA.
Step 5 From the Access Privileges drop-down menu, select ReadOnly.
Step 6 Check the Apply to All Nodes check box if the check box is active.
Step 7 Click Save.
Step 8 Click OK.

What to Do Next
For Basic Paging, go to Set Default Codec to G.711, on page 258
For Advanced Notification, go to Configure Media Resources for Legacy Paging, on page 257
Configure Media Resources for Legacy Paging

If you are implementing an Advanced Notification deployment that includes legacy paging devices, perform these tasks to configure media resources with an MTP so that InformaCast can send broadcasts to your legacy paging devices.

**Before You Begin**

Configure SNMP for Paging, on page 255

**Procedure**

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<td>Add your media resource group to a media resource group list.</td>
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</table>

Create a Media Resource Group

Perform this procedure if you are configuring an Advanced Notification deployment that connects to legacy paging devices. You must configure a media resource group that includes an Media Termination Point (MTP) resource. The MTP is required to connect the legacy paging interface.

**Procedure**

1. From Cisco Unified CM Administration, choose Media Resources > Media Resource Group.
2. Click Add New.
3. Enter a Name and Description for the group.
4. Use the arrows to move MTP from the Available Media Resources area to the Selected Media Resources area.
5. Click Save.

**What to Do Next**

Create a Media Resource Group List, on page 257

Create a Media Resource Group List

If you have configured a media resource group with an MTP to connect to a legacy paging interface, configure a media resource group that includes your media resource group.

**Before You Begin**

Create a Media Resource Group, on page 257
Procedure

**Step 1** From Cisco Unified CM Administration, choose **Media Resources > Media Resource Group List**.

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

**Step 3** Enter a **Name** and **Description** for the list.

**Step 4** Use the arrows to move the media resource group that you created to the **Selected Media Resource Groups** area.

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

**What to Do Next**

*Set Default Codec to G.711*, on page 258

**Configure Region for Paging**

For either Basic Paging or Advanced Notification deployments, you must set up a region for your paging deployment.

**Before You Begin**

For Basic Paging, *Configure SNMP for Paging*, on page 255

For Advanced Notification, *Configure Media Resources for Legacy Paging*, on page 257

**Procedure**

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<td><strong>Step 2</strong></td>
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**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 255
Procedure

Step 1 In Cisco Unified CM Administration, choose System > Region Information > Region.
Step 2 Click Add New.
Step 3 In the Name field, enter ICVA.
Step 4 Click Save.
Step 5 In the Regions text box, select all regions by pressing the CTRL key and clicking all of the selected regions.
Step 6 From the Maximum Audio Bit Rate drop-down list box, select 64 kbps (G.722, G.711).
Step 7 From the Maximum Session Bit Rate for Video Calls column click the None radio button.
Step 8 Click Save.

What to Do Next
Configure a Device Pool for Paging, on page 259

Configure a Device Pool for Paging

Perform this procedure for both Basic Paging and Advanced Notification deployments to configure a device pool for your paging deployment.

Before You Begin
Set Default Codec to G.711, on page 258

Procedure

Step 1 In Cisco Unified CM Administration, choose System > Device Pool.
Step 2 Click Add New.
Step 3 In the Device Pool Name field, enter ICVA.
Step 4 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.
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 If you have an Advanced Notification deployment that includes legacy paging devices, from the Media Resource Group List drop-down list box, select the media resource group list that you configured to include an MTP.
Step 8 From the SRST Reference drop-down list box, select Disable.
Step 9 Click Save.

What to Do Next
Configure Partitions and Calling Search Spaces for Paging, on page 260
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.
- For Advanced Notification deployments, create a single partition and CSS for InformaCast paging. If you are deploying CallAware, you must also configure a unique partition and CSS combination for each CallAware redirect.

Before You Begin
Configure Region for Paging, on page 258

Procedure

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Configure Route Partition for InformaCast Paging

Create a route partition for InformaCast paging in either a Basic Paging or Advanced Notification deployment.

Before You Begin
Configure a Device Pool for Paging, on page 259

Procedure

**Step 1** In Cisco Unified CM Administration, choose Call Routing > Class of Control > Route Partitions.
**Step 2** Click Add New.
**Step 3** In the Name field, enter the following name and description for the partition: ICVA-CTIOutbound, ICVA-Do not add to any phone CSS.
**Step 4** Click Save.
Configure Calling Search Space for InformaCast Paging

Perform this procedure to configure a calling search space for InformaCast paging in either a Basic Paging or Advanced Notification deployment.

Before You Begin
Configure Route Partition for InformaCast Paging, on page 260

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
For Basic Paging or Advanced Notification without CallAware, Configure CTI Ports for Paging, on page 265
For Advanced Notification with CallAware, Configure Route Partitions for CallAware, on page 261

Configure Route Partitions for CallAware

If you are deploying CallAware in an Advanced Notification paging deployment. perform this procedure to set up a unique partition for each CallAware redirect.

Before You Begin
Configure Calling Search Space for InformaCast Paging, on page 261
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 the following name and description for the partition: ICVA-Redirect1-CA, ICVA Redirect 1 CallAware.
Step 4  Click Save
Step 5  For each additional call redirect, create a new partition, incrementing the redirect number that you enter in the Name field. For example, ICVA-Redirect2-CA.

What to Do Next
Configure Calling Search Spaces for CallAware, on page 262

Configure Calling Search Spaces for CallAware

If you are deploying CallAware in an Advanced Notification deployment, you must configure a unique calling search space (CSS) for each CallAware redirect.

Before You Begin
Configure Route Partitions for CallAware, on page 261

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-Redirect1-CA.
Step 4  Click Save.
Step 5  Repeat the above steps until you have a separate CSS for each call redirect. For each CSS, increment the redirect number in the Name field. For example, ICVA-Redirect2-CA.

What to Do Next
Configure CTI Route Points for CallAware, on page 262

Configure CTI Route Points for CallAware

Perform this procedure only if you have an Advanced Notification paging deployment that includes CallAware. You must configure CTI route points for each CallAware redirect.

Before You Begin
Configure Partitions and Calling Search Spaces for Paging, on page 260
Procedure

Step 1 In Cisco Unified CM Administration, choose Device > CTI Route Point.
Step 2 Click Add New.
Step 3 In the Device Name field, enter a name for your route point. This name must match the corresponding route point name that you configure in the InformaCast Virtual Appliance.
Step 4 Enter a Description for the route point. For example, ICVA Redirect 1 CallAware.
Step 5 From the Device Pool drop-down list box, choose ICVA.
Step 6 From the Calling Search Space drop-down list box, choose the calling search spaces that you set up for this CallAware redirect.
Step 7 Click Save.
Step 8 In the Association area, click Line [1] - Add a new DN.
Step 9 In the Directory Number field, enter the directory number that CallAware will monitor.
Step 10 From the Route Partition drop-down list box, select the partition that you set up for this CallAware redirect.
Step 11 In the Call Forward and Call Pickup Settings area, leave the Forward All field empty.
Step 12 For each of the remaining call forward options, configure the following settings:
   a) In the Destination text box, enter the directory number to which you want to redirect calls.
   b) From the Calling Search Space drop-down list box, select the CSS that you created for this redirect.
   c) Click Save.
Step 13 Repeat this procedure for each CallAware redirect.

What to Do Next
Perform one of the following procedures to enable the Built in Bridge on the phone:

- Enable Built in Bridge for Cluster, on page 263
- Enable Built in Bridge for a Phone, on page 264

Enable Built in Bridge for Cluster
Perform this procedure if you have an Advanced Notification paging deployment that includes CallAware. When you use this procedure to enable the Built in Bridge using a clusterwide service parameter, 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 setting.

Before You Begin
Configure CTI Route Points for CallAware, on page 262
Procedure

Step 1  In 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**.

What to Do Next

Enable Built in Bridge for a Phone, on page 264

Enable Built in Bridge for a Phone

If you have an Advanced Notification paging deployment that uses CallAware, perform this procedure to enable an individual phone's Builtin in Bridge.

You can also use a service parameter to set the Builtin in Bridge clusterwide default setting to enabled. However, an individual phone's Builtin in Bridge setting in the **Phone Configuration** window overrides the clusterwide service parameter default.

Before You Begin

Enable Built in Bridge for Cluster, on page 263

Procedure

Step 1  In Cisco Unified CM Administration, choose **Device > Phone**.
Step 2  Click **Find**.
Step 3  Select the agent phone.
Step 4  From the **Built in Bridge** drop-down list, choose one of the following options:
  - **On**—The Builtin in Bridge is enabled.
  - **Off**—The Builtin in Bridge is disabled.
  - **Default**—The setting of the clusterwide **Builtin Bridge Enable** service parameter is used.
Step 5  Click **Save**.

What to Do Next

Configure CTI Ports for Paging, on page 265
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.
- For Advanced Notification deployments, you must create a minimum of two CTI ports for InformaCast paging and two CTI ports for CallAware.

Before You Begin

For Basic Paging, Configure Calling Search Space for InformaCast Paging, on page 261
For Advanced Notification, Enable Built in Bridge for a Phone, on page 264

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 or ICVA-CA-001 for a CallAware port.
Step 5 In the Description field, enter a description for the port. For example, InformaCast Recording Port or CallAware CTI 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 one of the following ports:
  - For InformaCast ports, select ICVA-CTIOutbound.
• For CallAware ports, select the CallAware partitions. For example, ICVA-Redirect1-CA.

Step 14 In the Display (Internal Caller ID) text box, enter InformaCast or CallAware depending on which type of port you are configuring.

Step 15 In the ASCII Display (Internal Caller ID) text box, enter InformaCast or CallAware depending on which type of port you are configuring.

Step 16 If you have an Advanced Notification deployment that includes CallAware, from the Monitoring Calling Search Space drop-down list box, select the CSS that you created for the CallAware redirect. For example, ICVA-Redirect1-CA.

Step 17 Click Save.

Step 18 Repeat this procedure for each CTI port that you need.

What to Do Next
For Basic Paging, go to Configure Access Control Group with AXL Access, on page 267
For Advanced Notification, go to Configure SIP Trunk for Legacy Paging Devices, on page 266

Configure SIP Trunk for Legacy Paging Devices
Perform this procedure if you have an Advanced Notification paging deployment and you want to connect to legacy paging devices. You must configure a SIP trunk to enable the Legacy Paging Interface.

Before You Begin
Configure CTI Ports for Paging, on page 265

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Trunk.
Step 2 Click Add New.
Step 3 From the Trunk Type drop-down list box, select SIP Trunk.
Step 4 Click Next.
Step 5 In the Device Name text box, enter ICVA.
Step 6 In the Description, field enter a description for the trunk.
Step 7 From the Device Pool drop-down menu, select ICVA.
Step 8 From the Calling Search Space drop-down list box, select ICVA.
Step 9 In the SIP Information area, enter the InformaCast Virtual Appliance IP address in the Destination Address text box.
Step 10 From the SIP Trunk Security Profile drop-down list box, select Non Secure SIP Trunk Profile.
Step 11 From the SIP Profile drop-down list box, select Standard SIP Profile.
Step 12 Click Save.
What to Do Next

Configure Access Control Group with AXL Access, on page 267

Configure Access Control Group with AXL Access

Perform this procedure for either Basic Paging or Advanced Notification deployments to create an access control group that includes AXL access.

Before You Begin

For Basic Paging, Configure CTI Ports for Paging, on page 265
For Advanced Notification, Configure SIP Trunk for Legacy Paging Devices, on page 266

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.
Step 10 Click Save.

What to Do Next

Configure Application User for Paging, on page 267

Configure Application User for Paging

Perform this procedure to configure an application user for either Basic Paging or Advanced Notification deployments as follows:

• For Basic Paging, configure an InformaCast application user.
• For Advanced Notification, if you are deploying the InformaCast, CallAware, and PushToTalk features, you must create a separate application user for each feature.

Before You Begin

Configure Access Control Group with AXL Access, on page 267
Procedure

Step 1  In Cisco Unified CM Administration, choose User Management > Application User.
Step 2  Click Add New.
Step 3  In the User ID text box, enter a user ID for the application user. For example, ICVA InformaCast, ICVA CallAware, or ICVA PushToTalk.
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 Call Monitoring—for CallAware or PushToTalk application users only
   • 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.
Step 11 If you have an Advanced Notification deployment, repeat all the steps in this procedure until you have configured application users for InformaCast, CallAware, and PushToTalk.

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 268
   • Enable Web Access for Common Phone Profile, on page 269

Enable Web Access for a Phone

Perform this procedure in Basic Paging or Advanced Notification deployments 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 269.

Before You Begin

Configure Application User for Paging, on page 267
Procedure

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

Step 2
Click Find and select the phone for which you want to enable web access.

Step 3
In the Product Specific Configuration Layout area, from the Web Access drop-down menu, select Enabled.

Step 4
Click Save.

What to Do Next
For Basic Paging or Advanced Notification, go to Configure Authentication URL, on page 269

Enable Web Access for Common Phone Profile

Perform this procedure in either Basic Paging or Advanced Notification deployments 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 268.

Before You Begin
Configure Application User for Paging, on page 267

Procedure

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

Step 2
Click Find and select the profile that applies to the group of phones for which you want to enable web access.

Step 3
In the Product Specific Configuration Layout area, from the Web Access drop-down list, select Enable.

Step 4
Click Save.

Step 5
Click Apply Config to reset the phones that use the Common Phone Profile.

Step 6
Click OK.

What to Do Next
For Basic Paging or Advanced Notification, go to Configure Authentication URL, on page 269

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

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<td><strong>Step 2</strong></td>
<td>Reset the phones in your deployment so that your phones use the new settings.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Verify that the phones in your deployment use the new authentication URL settings.</td>
</tr>
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</table>

### Set Authentication URL

Perform this procedure to set the Cisco Unified Communications Manager authentication URL to point to the InformaCast Virtual Appliance.

**Procedure**

1. In Cisco Unified CM Administration choose System > Enterprise Parameters.
2. Scroll to the Phone URL Parameters area, and in the URL Authentication field, enter `http://<IP Address>:8081/InformaCast/phone/auth` where `<IP Address>` is the IP Address of the InformaCast Virtual Appliance.
   - **Note**: Make a note of the existing URL in the URL Authentication field. You may need this when you configure InformaCast. See your InformaCast documentation for details.
3. Scroll to the Secured Phone URL Parameters area, and in the Secured Authentication URL field, enter `http://<IP Address>:8081/InformaCast/phone/auth` where `<IP Address>` is the IP Address of the InformaCast Virtual Appliance.
4. Click Save.

### What to Do Next

*Reset Your Phones, on page 270*

### 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 270

Procedure

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

**Step 2** In the From Phone Where box, select Device Pool.

**Step 3** Set the other drop-down menus and field items to settings that will bring up the device pools that you contain your phones.

**Step 4** Click Find.

**Step 5** Select the device pools that you want to reset.

**Step 6** Click Reset Selected.

**Step 7** Click Reset.

What to Do Next

Test Your Phones, on page 271

Test Your Phones

Verify that your phones are authenticating with the InformaCast Virtual Appliance.

Before You Begin

Reset Your Phones, on page 270

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.

For Advanced Notification, Configure PushToTalk Service Integration, on page 272.
Configure PushToTalk Service Integration

If you have already configured an Advanced Notification paging deployment in Cisco Unified Communications Manager, perform the following tasks to add PushToTalk services.

Before You Begin

If you have not done so, configure an application user specifically for PushToTalk. For details, see Configure Application User for Paging, on page 267

Procedure

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<td><strong>Step 7</strong></td>
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Create a PushToTalk Service Definition

If you want to enable PushToTalk services, you must create a service definition for PushToTalk to be able to display interactive content with text and graphics on Cisco Unified IP Phones.
Procedure

Step 1  In Cisco Unified CM Administration, choose Device > Device Settings > Phone Services.
Step 2  Click Add New.
Step 3  Enter a Service Name for the service. For example, PushToTalk.
Step 4  Enter a Service Description for the service. For example, Intercom Functionality.
Step 5  In the Service URL text box, enter http://<InformaCast Virtual Appliance IP Address>:8085/PushToTalk/PhoneMenu.action?sep=#DEVICENAME#.
Step 6  Check the Enable check box.
Step 7  Click Save.

What to Do Next
Perform the following procedures to configure the parameters by which PushToTalk can be customized for each phone:

- Configure PhoneGroup Parameter for PushToTalk, on page 273
- Configure SkipConfirm Parameter for PushToTalk, on page 274
- Configure Directory Number Parameter for PushToTalk, on page 275

Configure PhoneGroup Parameter for PushToTalk
Use this procedure to configure the PhoneGroup parameter, which allows you to limit a phone subscribed to this service’s parameter to only displaying a subset of the phone groups available.

Before You Begin
Create a PushToTalk Service Definition, on page 272
Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Device Settings > Phone Services.
Step 2 Click Find and select the service that you created for PushToTalk.
Step 3 Click the New Parameter button.
Step 4 In the Parameter Name text box, enter PhoneGroupIds.
Step 5 In the Parameter Display Name text box, enter Phone Group Default.
Step 6 In the Parameter Description text box, enter text similar to “If you wish to limit the phone to less than all phone groups, enter the phone group ID(s).”
Step 7 Uncheck the Parameter is Required check box.
Step 8 Uncheck the Parameter is a Password (mask contents) check box.
Step 9 Click Save.
Step 10 Close the Configure Cisco IP Phone Service Parameter window.
Step 11 Click Update Subscriptions.
Step 12 Click Save.

What to Do Next
Configure SkipConfirm Parameter for PushToTalk, on page 274

Configure SkipConfirm Parameter for PushToTalk
Use this procedure to configure the SkipConfirm parameter for PushToTalk. The SkipConfirm parameter allows you to (in conjunction with the PhoneGroupIds parameter) immediately enter a PushToTalk session by pressing the side button on wireless phones or the Services button on desktop phone.

Before You Begin
Configure PhoneGroup Parameter for PushToTalk, on page 273
Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Device Settings > Phone Services.
Step 2 Click Find and select the phone service that you created for PushToTalk.
Step 3 Click New Parameter.
Step 4 In the Parameter Name text box, enter SkipConfirm.
Step 5 In the Parameter Display Name text box, enter SkipConfirmation.
Step 6 In the Default Value text box, enter N.
Step 7 In the Parameter Description text box, enter text similar to, “Skip (Y/N) the confirmation page when limiting to only one specific PushToTalk phone group.”
Step 8 Uncheck the Parameter is Required check box.
Step 9 Uncheck the Parameter is a Password (mask contents) check box.
Step 10 Click Save.
Step 11 Close the Configure Cisco IP Phone Service Parameter window.
Step 12 Click the Update Subscriptions button.
Step 13 Click Save.

What to Do Next

Configure Directory Number Parameter for PushToTalk, on page 275

Configure Directory Number Parameter for PushToTalk

Use this procedure to configure the Directory Number parameter for PushToTalk services in an Advanced Notification deployment. The Directory Number parameter allows you to immediately enter a PushToTalk session by pressing the side button on wireless phones or the Services button on desktop phone.

Before You Begin

Configure SkipConfirm Parameter for PushToTalk, on page 274
Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Device Settings > Phone Services.
Step 2 Click Find and select the service that you set up for PushToTalk.
Step 3 Click the New Parameter button.
Step 4 In the Parameter Name text box, enter DN.
Step 5 In the Parameter Display Name text box, enter Directory Number.
Step 6 In the Parameter Description text box, enter text similar to "If skipping confirmation page when limiting to a specific phone group, use this directory number when starting a one-to-one or intercom session."
Step 7 Uncheck the Parameter is Required check box.
Step 8 Uncheck the Parameter is a Password (mask contents) check box.
Step 9 Click Save.
Step 10 Close the Configure Cisco IP Phone Service Parameter window.
Step 11 Click the Update Subscriptions button.
Step 12 Click Save.

What to Do Next
Assign Phones to PushToTalk Service, on page 276

Assign Phones to PushToTalk Service

Use this procedure to assign a phone to the PushToTalk service and to customize how the Phone Group, Skip Confirmation and Directory Number parameters are set when the user initiates a PushToTalk session.

Before You Begin
Configure Directory Number Parameter for PushToTalk, on page 275

Procedure

Step 1 In Cisco Unified CM Administration, choose Device > Phone.
Step 2 Click Find and select the phone that you want to use with PushToTalk. The Phone Configuration window opens.
Step 3 From the Related Links drop-down list box, select Subscribe/Unsubscribe and click Go. The Cisco IP Phone Services window opens.
Step 4 From the Select a Service drop-down list box, select PushToTalk and click Next.
Step 5 Leave the Service Name and ASCII Service Name fields with their default values.
Step 6 If you want to limit the phone to specific phone groups for PushToTalk, in the Phone Group Default text box, enter the phone group IDs that you want to include.
Note If you leave the field empty, the Phone Groups menu appears whenever you initiate a PushToTalk session.
Step 7 If you want the phone to skip the confirmation screen when entering a PushToTalk session, enter Y in the Skip Confirmation field.
If you enter N, the Start confirmation appears whenever you initiate a PushToTalk session.

**Note**

If you want the phone to enter a one-to-one or intercom session immediately, in the **Directory Number** text box, enter the directory number for the one-to-one or intercom session.

**Note**

If you leave the field empty, you must enter the directory number in the phone interface when you initiate a one-to-one or intercom PushToTalk session.

**Step 8**

If you want the phone to enter a one-to-one or intercom session immediately, in the **Directory Number** text box, enter the directory number for the one-to-one or intercom session.

**Note**

If you leave the field empty, you must enter the directory number in the phone interface when you initiate a one-to-one or intercom PushToTalk session.

**Step 9**

Click **Subscribe**.

**Step 10**

Close the **Cisco IP Phone Services** window.

**Step 11**

Click **Reset**.

The **Device Reset** window appears.

**Step 12**

Click **Reset**.

---

**What to Do Next**

[Configure Wireless Phones for PushToTalk](#), on page 277

---

**Configure Wireless Phones for PushToTalk**

Use this procedure to assign PushToTalk services to a wireless phone and to configure how the Phone Group, Skip Confirmation and Directory Number parameters are set when PushToTalk feature is invoked from a wireless phone.

**Before You Begin**

Assign Phones to PushToTalk Service, on page 276

**Procedure**

**Step 1**

In Cisco Unified CM Administration, choose **Device > Phone**.

**Step 2**

Click **Find** and select the wireless phone.

The **Phone Configuration** window appears.

**Step 3**

In the **Application URL** text box, enter a URL to configure the Phone Group, Skip Confirmation and Directory Number parameters and control how PushToTalk behaves when the feature is invoked from the phone. Refer to the following table for the URLs.

<table>
<thead>
<tr>
<th>Desired Behavior for PushToTalk</th>
<th>URL to Enter</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phone Group menu prompts user to select a phone group.</td>
<td>http://&lt;InformaCast IP Address&gt;:8085/PushToTalk/PhoneMenu.action?sep=#DEVICENAME#</td>
</tr>
<tr>
<td>Bypass the Phone Group menu. The phone group is limited to the phone groups in the URL.</td>
<td>http://&lt;InformaCast IP Address&gt;:8085/PushToTalk/PhoneMenu.action?sep=#DEVICENAME#;PhoneGroupId=x where x represents the phone group. For example, PhoneGroupId=3,5.</td>
</tr>
<tr>
<td>Desired Behavior for PushToTalk</td>
<td>URL to Enter</td>
</tr>
<tr>
<td>--------------------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>Bypass Phone Group menu and bypass the confirmation screen for a standard PushToTalk session.</td>
<td>http://&lt;InformaCast IP Address&gt;:8085/PushToTalk/PhoneMenu.action?sep=#DEVICENAME#;PhoneGroupIds=x;SkipConfirm=Y where x represents the phone group. For example, PhoneGroupIds=3,5</td>
</tr>
<tr>
<td>Bypass Phone Group menu, confirmation screen, and directory number selection for a one-to-one or intercom PushToTalk session.</td>
<td>http://&lt;InformaCast IP Address&gt;:8085/PushToTalk/PhoneMenu.action?sep=#DEVICENAME#;PhoneGroupIds=x;SkipConfirm=Y;DN=xx where x represents the phone group and xx represents the directory number for the one-to-one or intercom PushToTalk session. For example, PhoneGroupIds=3,5;SkipConfirm=Y;DN=2004</td>
</tr>
</tbody>
</table>

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

**Step 5** Click **Reset**.
The Device Reset window appears.

**Step 6** Click **Reset**.

**What to Do Next**

Install Audio Files for PushToTalk, on page 278

**Install Audio Files for PushToTalk**

Perform this procedure to install the following audio files that PushToTalk uses to indicate session activity:

- PTT_start.raw
- PTT_accept.raw
- PTT_change.raw
- PTT_end.raw
- PTT_active.raw

**Before You Begin**

Perform one of the following procedures to set up the PushToTalk service on your phones:

- Assign Phones to PushToTalk Service, on page 276
- Configure Wireless Phones for PushToTalk, on page 277
Procedure

Step 1  Use an SFTP client to access the five PushToTalk audio files from the following default location on the InformaCast Virtual Appliance: /usr/local/singlewire/PushToTalk/web/sounds.
Step 2  Use the SFTP client to download the files to a location that the Cisco Unified Communications Manager TFTP server can access.
Step 3  Log in to Cisco Unified Operating System Administration and choose Software Upgrades > TFTP File Management.
Step 4  Click Upload File.
Step 5  Browse to the location where you saved the PushToTalk audio files.
Step 6  For each audio file, perform the following steps:
   a) Select the audio file.
   b) In the Directory field, enter the subdirectory where the file will reside.
   c) Click Upload File.
Step 7  Restart the TFTP service by performing the following steps:
   a) From Cisco Unified Serviceability, choose Tools > Control Center - Feature Services.
   b) From the Server drop-down list box, select the server on which the TFTP service is running.
   c) Click the Cisco Tftp radio button.
   d) Click the Restart button.
   e) Click OK.

What to Do Next

Optional. Assign a Directory URI to a Phone, on page 279

Assign a Directory URI to a Phone

Use this procedure if you want to assign a directory URI to a phone, thereby allowing you to use URI dialing for that phone.

Procedure

Step 1  In Cisco Unified CM Administration, choose Device > Phone.
Step 2  Click Find and select the phone for which you want to assign a directory URI. The Phone Configuration window appears.
Step 3  In the Association Information pane that appears on the left, click the phone line. The Directory Number Configuration window appears.
Step 4  In the Directory URIs section, enter a directory URI in the URI text box.
Step 5  Click Save.
Intercom

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**

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Intercom Partition, on page 282</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 284</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 285</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 290</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 295</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 7

**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 the Add New button.

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, Cisco 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 Related Topics section 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 284

**Related Topics**

Intercom Partition Configuration Fields, on page 283

---

**Intercom Partition Configuration Fields**

**Table 9: Intercom Partition Configuration Fields**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>The name of the intercom partition that you selected displays in this box.</td>
</tr>
<tr>
<td>Description</td>
<td>If you entered a description of the intercom partition that you selected, it is displayed here. If you did not enter a description when you added the Intercom partition, you can add it now.</td>
</tr>
<tr>
<td>Time Schedule</td>
<td>The drop-down list is populated with time schedules that you can add from <strong>Call Routing &gt; Class of Control &gt; Time Schedule</strong>.</td>
</tr>
<tr>
<td>Time Zone</td>
<td>• If you want the time zone to be the same as the originating device, click the radio button next to Originating Device.</td>
</tr>
<tr>
<td></td>
<td>• If you want to set a specific time zone, click the Specific Time Zone radio button and select the correct time zone from the drop-down list.</td>
</tr>
</tbody>
</table>
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 Related Topics section for more information about the fields and their configuration options.

**Step 4** Click Save.

**What to Do Next**

Configure an Intercom Translation Pattern, on page 285

**Related Topics**

Intercom Calling Search Space Configuration Fields, on page 284

### Intercom Calling Search Space Configuration Fields

#### Table 10: Intercom Calling Search Space Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>Enter a name in the Intercom Calling Search Space Name field. The name can comprise up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_). Ensure each calling search space name is unique to the system.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Use concise and descriptive names for your intercom calling search spaces. The CompanynameLocationCalltype format usually provides a sufficient level of detail and is short enough to enable you to quickly and easily identify a calling search space. For example, CiscoDallasMetroCS identifies a calling search space for toll-free, inter-local access and transport area (LATA) calls from the Cisco office in Dallas.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description in the Description field. The description can include up to 50 characters in any language, and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_), but it cannot include double-quotes (&quot;), percentage sign (%), ampersand (&amp;), or angle brackets (&lt;&gt;).</td>
</tr>
</tbody>
</table>

Intercom Route Partitions for this Calling Search Space
Choose an intercom partition in the Available Intercom Partitions list box and add it to the Selected Intercom Partitions list box by clicking the arrow button between the two list boxes.

To add a range of intercom partitions at once, click the first intercom partition in the range; then, hold down the Shift key while clicking the last intercom partition in the range. Click the arrow button between the two list boxes to add the range of partitions.

To add multiple intercom partitions that are not contiguous, hold down the Control (Ctrl) key while clicking multiple intercom partitions. Click the arrow button between the two list boxes to add the chosen intercom partitions.

**Note**  The length of the intercom partition names limits the maximum number of intercom partitions that can be added to an intercom calling search space. Configure Intercom Partition, on page 282 provides examples of the maximum number of partitions that can be added to an intercom calling search space if intercom partition names are of fixed length.

Selected Intercom Partitions (Ordered by highest priority)  To change the priority of an intercom partition, choose an intercom partition name in the Selected Intercom Partitions list box. Move the intercom partition up or down in the list by clicking the arrows on the right side of the list.

---

### 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 the Copy button beside 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. See the Related Topics section for more information about the fields and their configuration options.

**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.
Table 11: Translation Pattern Configuration Field Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pattern Definition</td>
<td>Enter the Intercom Translation Pattern, including numbers and wildcards (do not use spaces), in the Intercom Translation Pattern field. 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 +. If you leave this field blank, you must select a partition from the Partition drop-down list box. Note Ensure that the Intercom Translation Pattern, which uses the chosen intercom partition, route filter, and numbering plan combination, is unique. Check the route pattern/hunt pilot, translation pattern, directory number, call park number, call pickup number, or meet-me number if you receive a message that indicates duplicate entries. Alternatively, check the route plan report if you receive a message that indicates duplicate entries.</td>
</tr>
<tr>
<td>Intercom Translation Pattern</td>
<td>Choose an intercom partition. If you do not want to assign an intercom partition, choose &lt;None&gt;. If you choose &lt;None&gt;, you must enter a value in the Intercom Translation Pattern field. You can configure the number of intercom partitions that display in this drop-down list by using the Max List Box Items enterprise parameter. If more intercom partitions exist than the Max List Box Items enterprise parameter specifies, the Find button displays next to the drop-down list. Click the Find button to display the Find and List Partitions window. Find and choose an intercom partition name. Note To set the maximum list items, choose System &gt; Enterprise Parameters and choose CCMAdmin Parameters. Note Make sure that the combination of intercom translation pattern, route filter, and intercom partition is unique within the Cisco Unified Communications Manager cluster.</td>
</tr>
<tr>
<td>Partition</td>
<td>Enter a description for the Intercom 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>Field</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>Choose a numbering plan. If your Intercom Translation Pattern includes the @ wildcard, you may choose a numbering plan. The optional act of choosing a numbering plan restricts certain number patterns.</td>
</tr>
<tr>
<td>Route Filter</td>
<td>Choosing an optional route filter restricts certain number patterns. See topics related to wildcards and special characters in route patterns and hunt pilots. The route filters that display depend on the numbering plan that you choose from the Numbering Plan drop-down list.</td>
</tr>
<tr>
<td></td>
<td>If more than 250 route filters exist, the Find button displays next to the drop-down list box. Click the Find button to display the Select Route Filters window. Enter a partial route filter name in the List items where Name contains field. Click the desired route filter name in the list of route filters that displays in the Select item to use box and click Add Selected.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> To set the maximum list box items, choose System &gt; Enterprise Parameters and choose CCMAadmin Parameters.</td>
</tr>
</tbody>
</table>
| MLPP Precedence     | Choose an MLPP precedence setting for this intercom translation pattern from the drop-down list:  
|                     | • Executive Override - Highest precedence setting for MLPP calls.  
|                     | • Flash Override - Second highest precedence setting for MLPP calls.  
|                     | • Flash - Third highest precedence setting for MLPP calls.  
|                     | • Immediate - Fourth highest precedence setting for MLPP calls.  
|                     | • Priority - Fifth highest precedence setting for MLPP calls.  
|                     | • Routine - Lowest precedence setting for MLPP calls.  
|                     | • Default - Does not override the incoming precedence level but rather lets it pass unchanged.  
<p>| Calling Search Space| From the drop-down list, choose the Intercom Calling Search Space for which you are adding an Intercom Translation Pattern, if necessary. You can configure the number of Intercom Calling Search Spaces that display in this drop-down list by using the Max List Box Items enterprise parameter. If more Intercom Calling Search Spaces exist than the Max List Box Items enterprise parameter specifies, the Find button displays next to the drop-down list. Click the Find button to display the Find and List Calling Search Space window. |</p>
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Route Option                | The Route Option designation indicates whether you want this intercom translation pattern to be used for routing calls (such as 9.@ or 8[2-9]XX) or for blocking calls. Choose the Route this pattern or Block this pattern radio button. If you choose the Block this pattern radio button, you must choose the reason for which you want this intercom translation pattern to block calls. Choose a value from the drop-down list:  
  • No Error  
  • Unallocated Number  
  • Call Rejected  
  • Number Changed  
  • Invalid Number Format  
  • Precedence Level Exceeded |
| Provide Outside Dial Tone   | Outside dial tone indicates that Cisco Unified Communications Manager routes the calls off the local network. Check this check box for each Intercom Translation Pattern that you consider to be off network. |
| Urgent Priority             | If the dial plan contains overlapping patterns, Cisco Unified Communications Manager does not route the call until the interdigit timer expires (even if it is possible to dial a sequence of digits to choose a current match). Check this check box to interrupt interdigit timing when Cisco Unified Communications Manager must route a call immediately. By default, the Urgent Priority check box displays as checked. Unless your dial plan contains overlapping patterns or variable length patterns that contain !, Cisco recommends that you do not uncheck the check box. |

**Calling Party Transformations**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party's External Phone Number Mask</td>
<td>Check the check box if you want the full, external phone number to be used for calling line identification (CLID) on outgoing calls.</td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td>Enter a transformation mask value. Valid entries include the digits 0 through 9, the wildcard characters asterisk (*) and octothorpe (#), the international escape character + and blank. If this field is blank and the preceding field is not checked, no calling party transformation takes place.</td>
</tr>
</tbody>
</table>
| Prefix Digits (Outgoing Calls) | Enter prefix digits. Valid entries include the digits 0 through 9, the wildcard characters asterisk (*) and octothorpe (#), and the international escape character +.  
**Note** The appended prefix digit does not affect which directory numbers route to the assigned device. |
<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line ID Presentation</td>
<td>Cisco Unified Communications Manager uses calling line ID presentation (CLIP/CLIR) as a supplementary service to allow or restrict the originating caller phone number on a call-by-call basis.</td>
</tr>
<tr>
<td></td>
<td>Choose whether you want the Cisco Unified Communications Manager to allow or restrict the display of the calling party phone number on the called party phone display for this intercom translation pattern.</td>
</tr>
<tr>
<td></td>
<td>Choose Default if you do not want to change calling line ID presentation. Choose Allowed if you want Cisco Unified Communications Manager to allow the display of the calling number. Choose Restricted if you want Cisco Unified Communications Manager to block the display of the calling number.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> Use this parameter and the Connected Line ID Presentation parameter, in combination with the Ignore Presentation Indicators (internal calls only) device-level parameter, to configure call display restrictions. Together, these settings allow you to selectively present or restrict calling and/or connected line display information for each call. See the Cisco Unified Communications Manager Administration Guide for information about the Ignore Presentation Indicators (internal calls only) field.</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Cisco Unified Communications Manager uses calling name presentation (CNIP/CNIR) as a supplementary service to allow or restrict the originating caller name on a call-by-call basis.</td>
</tr>
<tr>
<td></td>
<td>Choose whether you want the Cisco Unified Communications Manager to allow or restrict the display of the calling party name on the called party phone display for this Intercom Translation Pattern.</td>
</tr>
<tr>
<td></td>
<td>Choose Default if you do not want to change calling name presentation. Choose Allowed if you want Cisco Unified Communications Manager to display the calling name information. Choose Restricted if you want Cisco Unified Communications Manager to block the display of the calling name information.</td>
</tr>
<tr>
<td>Connected Party Transformations</td>
<td></td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Cisco Unified Communications Manager uses connected line ID presentation (COLP/COLR) as a supplementary service to allow or restrict the called party phone number on a call-by-call basis.</td>
</tr>
<tr>
<td></td>
<td>Choose whether you want Cisco Unified Communications Manager to allow or restrict the display of the connected party phone number on the calling party phone display for this Intercom Translation Pattern.</td>
</tr>
<tr>
<td></td>
<td>Choose Default if you do not want to change the connected line ID presentation. Choose Allowed if you want to display the connected party phone number. Choose Restricted if you want Cisco Unified Communications Manager to block the display of the connected party phone number.</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager uses connected name presentation (CONP/CONR) as a supplementary service to allow or restrict the called party name on a call-by-call basis. Choose whether you want Cisco Unified Communications Manager to allow or restrict the display of the connected party name on the calling party phone display for this Intercom Translation Pattern. Choose Default if you do not want to change the connected name presentation. Choose Allowed if you want to display the connected party name. Choose Restricted if you want Cisco Unified Communications Manager to block the display of the connected party name.

### Called Party Transformations

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Name Presentation</td>
<td>Cisco Unified Communications Manager uses connected name presentation (CONP/CONR) as a supplementary service to allow or restrict the called party name on a call-by-call basis. Choose whether you want Cisco Unified Communications Manager to allow or restrict the display of the connected party name on the calling party phone display for this Intercom Translation Pattern. Choose Default if you do not want to change the connected name presentation. Choose Allowed if you want to display the connected party name. Choose Restricted if you want Cisco Unified Communications Manager to block the display of the connected party name.</td>
</tr>
</tbody>
</table>

#### Discard Digits
- Choose the discard digits instructions that you want to be associated with this intercom translation pattern. See the Cisco Unified Communications Manager System Guide for more information.
- **Note** The discard digits that display depend on the numbering plan that you choose from the Numbering Plan drop-down list.

#### Called Party Transform Mask
- Enter a transformation mask value. Valid entries include the digits 0 through 9, the wildcard characters asterisk (*) and octothorpe (#), the international escape character + and blank. If the field is blank, no transformation takes place. The dialed digits get sent exactly as dialed.

#### Prefix Digits (Outgoing Calls)
- Enter prefix digits. Valid entries include the digits 0 through 9, the wildcard characters asterisk (*) and octothorpe (#), the international escape character + and blank.
- **Note** The appended prefix digit does not affect which directory numbers route to the assigned device.

### 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 the **Add New** button.
- 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. See the Related Topics section for more information about the fields and their configuration options.

**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.

---

**What to Do Next**

Intercom Line and Speed Dial Configuration, on page 295

**Related Topics**

Intercom Directory Number Configuration Fields, on page 291

**Intercom Directory Number Configuration Fields**

The following table describes the fields that are available in the Intercom Directory Number Configuration window.

**Table 12: Intercom Directory Number Configuration Settings**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Intercom Directory Number Information</td>
<td>Enter a dialable phone number. Values can include numeric characters and route pattern wildcards and special characters except for (,) and (@). The intercom directory number that you enter can appear in more than one intercom partition. At the beginning of the intercom directory number, enter + if you want to use the international escape character +. For this field, + does not represent a wildcard; instead, entering + represents a dialed digit.</td>
</tr>
<tr>
<td>Intercom Directory Number</td>
<td></td>
</tr>
</tbody>
</table>
### Field | Description
--- | ---
Route Partition | Choose the intercom partition to which the intercom directory number belongs. Make sure that the intercom directory number that you enter in the Intercom Directory Number field is unique within the intercom partition that you choose. You can configure the number of intercom partitions that display in this drop-down list by using the Max List Items enterprise parameter. If more intercom partitions exist than the Max List Items enterprise parameter specifies, the Find button displays next to the drop-down list. Click the Find button to display the Find and List Partition window. Enter a partial intercom partition name in the List items where Name contains field. Click the desired intercom partition name in the list of intercom partitions that displays in the Select item to use box and click Add Selected. **Note** To set the maximum list box items, choose System > Enterprise Parameters > CCMAdmin Parameters.
Description | Enter a description of the intercom directory number and intercom route partition. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), or angle brackets (<>).
Alerting Name | Enter a name that you want to display on the phone of the caller. This setting, which supports the Identification Services for the QSIG protocol, applies to shared and nonshared directory numbers. If you configure an alerting name for a directory number with shared-line appearances, when the phone rings at the terminating PINX, the system performs the following tasks:

- Forwards the name of the caller that is assigned to the directory number.
- Applies the Connected Name Restrictions (CONR) that are configured for the translation pattern (if restrictions exist); the originating PINX may modify the CONR, depending on the route pattern configuration.

If you do not configure an alerting name, "Name Not Available" may display on the caller phone. If you do not enter a name for the Display (Internal Caller ID) field, the information in the Alerting Name field displays in the Display (Internal Caller ID) field.

If you set the Always Display Original Dialed Number service parameter to True, the alerting name does not display on the calling phone; only the original dialed number displays.

ASCII Alerting Name | This field provides the same information as the Alerting Name field, but you must limit input to ASCII characters. Devices that do not support Unicode (internationalized) characters display the content of the Alerting Name ASCII field.
Allow Control of Device from CTI | Check this check box to allow CTI to control and monitor a line on a device with which this intercom directory number is associated.
### Intercom Configuration Task Flow

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Associated Devices  | After you associate this intercom directory number with a device, this pane displays the device with which this intercom directory number is associated.  

**Note**  
An intercom directory number can be associated with at most one device.  
To edit a device with which this intercom directory number is associated, choose a device name in the Associated Devices pane and click the Edit Device button. The Phone Configuration window or Device Profile Configuration window displays for the device that you choose.  
To edit a line appearance that has been defined for this intercom directory number, choose a device name in the Associated Devices pane and click the Edit Line Appearance button. The Directory Number Configuration window or Device Profile Configuration window refreshes to show the line appearance for this DN on the device that you choose.  
To associate a device to this intercom directory number from the list of devices in the Dissociate Devices pane, choose a device in the Dissociate Devices pane and add it to the Associated Devices pane by clicking the up arrow between the two panes. |
| Dissociate Devices  | If you choose to dissociate an intercom directory number from a device, this pane displays the device(s) from which you dissociate this intercom directory number.  
Choose a device in the Associated Devices pane and add it to the Dissociate Devices pane by clicking the down arrow between the two panes. |
| Intercom Directory Number Settings |                                                                                                                                                                                                                   |
Calling Search Space

From the drop-down list, choose the appropriate intercom calling search space. An intercom calling search space comprises a collection of intercom partitions that are searched for numbers that are called from this intercom directory number. The value that you choose applies to all devices that are using this intercom directory number.

Changes result in an update of the numbers that the Call Pickup Group field lists.

You can configure calling search space for forward all, forward busy, forward no answer, forward no coverage, and forward on CTI failure directory numbers. The value that you choose applies to all devices that are using this directory number.

You must configure either primary forward all calling search space or secondary forward all calling search space or both for call forward all to work properly. The system uses these concatenated fields (Primary CFA CSS + Secondary CFA CSS) to validate the CFA destination and forward the call to the CFA destination.

Note

If the system is using partitions and calling search spaces, Cisco recommends that you configure the other call forward calling search spaces as well. When a call is forwarded or redirected to the call forward destination, the configured call forward calling search space gets used to forward the call. If the forward calling search space is None, the forward operation may fail if the system is using 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 may fail.

When you forward calls by using the CFwdAll softkey on the phone, the automatic combination of the line CSS and device CSS does not get used. Only the configured Primary CFA CSS and Secondary CFA CSS get used. If both of these fields are None, the combination results in two null partitions, which may cause the operation to fail.

If you want to restrict users from forwarding calls on their phones, you must choose a restrictive calling search space from the Forward All Calling Search Space field.

BLF Presence Group

Configure this field with the BLF presence group feature.

From the drop-down list box, choose a BLF Presence Group for this intercom directory number. The selected group specifies the devices, end users, and application users that can monitor this intercom directory number.

The default value for BLF Presence Group specifies Standard Presence group, configured with installation. BLF Presence groups that are configured in Cisco Unified Communications Manager Administration also appear in the drop-down list.

Presence authorization works with BLF presence groups to allow or block presence requests between groups.
Choose one of the following options to activate the auto answer feature for this intercom directory number:

- Auto Answer with Headset
- Auto Answer with Speakerphone

Note: Make sure that the headset or speakerphone is not disabled when you choose Auto Answer with headset or Auto Answer with speakerphone.

Note: Do not configure auto answer for devices that have shared lines.

Note: For an intercom line on a CTIPort device, autoanswer-speakerphone and autoanswer-headset means that the autoanswer is on. The speakerphone or headset options do not apply to CTIPort devices; instead, it just indicates that the line is capable of auto-answering. Applications have responsibility for terminating the media on CTIPort devices and can terminate the media on either type of output device.

From the drop-down list, choose a default activated device for this intercom directory number. The selected device specifies the phone on which this intercom directory number is activated by default. The drop-down list lists only devices that support intercom.

Note: You must specify a default activated device for this intercom directory number to be active as an intercom line.

Note: If an intercom directory number is specified in a device profile that is configured for Cisco Extension Mobility, that intercom directory number will display as an intercom line only when a user logs in to the specified default activated device by using that device profile, as long as the device supports the intercom feature.

### Intercom Line and Speed Dial Configuration

#### 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 <em>Cisco Unified Communications Manager Bulk Administration Guide</em> 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>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Assistant</td>
<td>With the Cisco Unified Communications Manager Assistant Configuration Wizard, <strong>Cisco Unified Communications Manager Assistant</strong> 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. 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>
<tr>
<td>Internet Protocol Version 6 (IPv6)</td>
<td>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.</td>
</tr>
<tr>
<td>Intercom directory numbers (lines)</td>
<td>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.</td>
</tr>
</tbody>
</table>
### Feature Configuration Guide for Cisco Unified Communications Manager, Release 10.5(2)

#### 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 Pickup</td>
<td>The call pickup groups do not include intercom calls.</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.
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 gsmfsmIsm 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, page 305
- Call Forwarding, page 309
- Call Pickup, page 325
- Call Park and Directed Call Park, page 347
- Extension Mobility, page 371
- Extension Mobility Cross Cluster, page 387
- Hold Reversion, page 413
- Accessing Hunt Groups, page 421
- Call Transfer, page 431
- External Call Transfer Restrictions, page 445
Prime Line Support

- Prime Line Support Overview, page 305
- Prime Line Support Prerequisites, page 305
- Prime Line Support Configuration Task Flow, page 305
- Prime Line Support Interactions, page 307
- Prime Line Support Troubleshooting, page 308

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.
Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Clusterwide Prime Line Support, on page 306</td>
<td>(Optional) Configure the Prime Line Support feature for the Cisco CallManager service, which applies to the entire cluster.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Prime Line Support for Devices, on page 307</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>
</tbody>
</table>

**Note** 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.

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.
Configure Prime Line Support for Devices

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 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.</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 Auto Answer drop-down list in Cisco Unified CM Administration, the Auto Answer configuration overrides the configuration for the Always Use Prime Line 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, page 309
- Call Forwarding Configuration Task Flow, page 311
- Call Forwarding Interactions and Restrictions, page 319

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**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>Step 1</td>
<td>Configure Partitions for Call Forwarding, on page 311</td>
<td>Administrators can configure partitions to restrict Call Forwarding to certain numbers based on the design criteria and requirements.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Calling Search Space for Call Forwarding, on page 313</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>Step 3</td>
<td>Configure Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires, on page 313</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>Step 4</td>
<td>Configure Call Forward No Bandwidth, on page 316</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>Step 5</td>
<td>Configure Call Forward Alternate Destination, on page 317</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>Step 6</td>
<td>Configure Other Call Forwarding Types, on page 318</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>Step 7</td>
<td>Enable Destination Override for Call Forwarding, on page 318</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 In Cisco Unified Communications Manager 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 Related Topics section 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 timeschedule to associate with this partition. The timeschedule 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.

- **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.

Step 8 Click Save.

Related Topics

Partition Name Guidelines for Call Forwarding, on page 312

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.

<table>
<thead>
<tr>
<th>Partition Name Length</th>
<th>Maximum Number of Partitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>2 characters</td>
<td>170</td>
</tr>
<tr>
<td>3 characters</td>
<td>128</td>
</tr>
<tr>
<td>4 characters</td>
<td>102</td>
</tr>
</tbody>
</table>
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.

Procedure

Step 1  From Cisco Unified CM Administration, select 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.

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.

**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**.

**Related Topics**

*Hunt Call Treatment Fields for Call Forwarding*, on page 314

**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>Note Forward Hunt No Answer or Forward Hunt Busy fields are designed to move calls through the route list. Queuing is used to hold callers in a route list. Therefore, if 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 are enabled, queuing is automatically disabled.</td>
</tr>
<tr>
<td>Field</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<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>• <strong>Do Not Forward Unanswered Calls</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Use Forward Settings of Line Group Member</strong> (replaces Use Personal Preferences check box)</td>
</tr>
<tr>
<td></td>
<td>• <strong>Forward Unanswered Calls to</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Destination</strong>—Enter a directory number to which calls must be forwarded to.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Calling Search Space</strong>—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>• <strong>Maximum Hunt Timer</strong>—Enter a value (in seconds) that specifies the maximum time for hunting without queuing.</td>
</tr>
<tr>
<td></td>
<td>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>
<tr>
<td></td>
<td><strong>Note</strong> If hunting exceeds the number of hops that the <strong>Forward Maximum Hop Count</strong> service parameter specifies, hunting expires before the 30 minute maximum hunt timer value, and the caller receives a reorder tone.</td>
</tr>
<tr>
<td>Forward Hunt Busy</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>• <strong>Do Not Forward Unanswered Calls</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Use Forward Settings of Line Group Member</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Forward Unanswered Calls to</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Destination</strong>—Enter a directory number to which calls must be forwarded to.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Calling Search Space</strong>—Choose a calling search space from the drop-down list which applies to all devices that use this directory number.</td>
</tr>
</tbody>
</table>
Configure Call Forward No Bandwidth

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Call Routing &gt; Directory Number Configuration. The Find and List Directory Numbers window is displayed.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click Find. A list of configured directory numbers is displayed.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose the directory number for which you want to configure call forward when there is insufficient bandwidth. The Directory Number Configuration window is displayed.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Configure the fields in the AAR Settings area. See the Related Topics section for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Related Topics

Directory Number Configuration Fields for Call Forwarding, on page 316

Directory Number Configuration Fields for Call Forwarding

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Mail</td>
<td>Check this check box to forward the call to the voicemail.</td>
</tr>
<tr>
<td></td>
<td>Note: When you check this check box, Cisco Unified Communications Manager ignores the values in the Destination and Calling Search Space fields.</td>
</tr>
<tr>
<td>AAR Destination Mask</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>AAR Group</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 Precedence Alternate Party Timeout service parameter from System &gt; Service Parameters.</td>
</tr>
</tbody>
</table>
Configure Call Forward Alternate Destination

**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**.

**Related Topics**

MLPP Alternate Party And Confidential Access Level Settings Fields for Call Forwarding, on page 317

**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>Retain this destination in the call forwarding history</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>
<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>
</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.

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.

Related Topics

Call Forwarding Fields

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.
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 <strong>True</strong>, 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 <strong>Call Diversion Hop Count</strong> service parameter that supports External Call Control, and the <strong>Call Forward Call Hop Count</strong> service parameter that supports Call Forwarding are independent of each other; they work separately.</td>
</tr>
</tbody>
</table>
## Call Forwarding Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<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>
<tr>
<td>Immediate Divert</td>
<td>When the Forward No Answer field in the Directory Number Configuration window is not configured, Call Forward uses the clusterwide CFNA timer service parameter, <strong>Forward No Answer Timer</strong>. 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).</td>
</tr>
<tr>
<td>Logical Partitioning</td>
<td>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.</td>
</tr>
<tr>
<td>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Multilevel Precedence and Preemption (MLPP)</td>
<td><strong>Call Forward Busy</strong></td>
</tr>
<tr>
<td></td>
<td>• You can optionally configure a preconfigured Precedence Alternate Party target for any MLPP-enabled station.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• The system preserves precedence of calls across multiple forwarded calls.</td>
</tr>
<tr>
<td></td>
<td>• 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 preemting call when the station goes off hook.</td>
</tr>
<tr>
<td></td>
<td><strong>Call Forward No Answer</strong></td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• Normally, precedence calls are routed to users and not to the voicemail system. The administrator sets the <strong>Use Standard VM Handling For Precedence Calls</strong> enterprise parameter to avoid routing precedence calls to voicemail systems.</td>
</tr>
</tbody>
</table>

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.
### Call Forwarding Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Originally called party name in</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>
<tr>
<td>Placed Call History</td>
<td></td>
</tr>
<tr>
<td>Secure Tone</td>
<td>Call Forward All is supported on protected phones.</td>
</tr>
<tr>
<td>Session Handoff</td>
<td>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.</td>
</tr>
</tbody>
</table>

### 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>
<tr>
<td>Feature</td>
<td>Restriction</td>
</tr>
<tr>
<td>-------------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Immediate Divert</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>Intercom</td>
<td>You cannot forward Intercom calls.</td>
</tr>
<tr>
<td>Log Out of Hunt Group</td>
<td>When a phone that is running SIP (7906, 7911, 7941, 7961, 7970, and 7971) 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.</td>
</tr>
<tr>
<td>Logical Partitioning</td>
<td>Logical partitioning handling does not take place in the following circumstances:</td>
</tr>
<tr>
<td></td>
<td>• When both the caller and forwarded devices are Voice over IP (VoIP) phones.</td>
</tr>
<tr>
<td></td>
<td>• When geolocation or a geolocation filter is not associated with any device.</td>
</tr>
<tr>
<td>Multilevel Precedence and Preemption (MLPP)</td>
<td>Multilevel Precedence and Preemption (MLPP) support for supplementary services specifies the following restrictions for Call Forwarding:</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
</tbody>
</table>
Call Pickup

- Call Pickup Overview, page 325
- Call Pickup Configuration Task Flow, page 327
- Call Pickup Interactions and Restrictions, page 343

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 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.
The longest alerting call (longest ringing time) gets picked up first if multiple incoming calls occur in that group. For other group call pickup, priority takes precedence over the ringing time if multiple associated pickup groups are configured.

**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</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 1 | Configure a Call Pickup Group, on page 329 | Configure a call pickup group for each of the call pickup features that you want to use:  
• Call Pickup  
• Group Call Pickup  
• Other Call Pickup  
• Directed Call Pickup  
• BLF Call Pickup  
You must define groups with unique names and numbers. |
| Step 2 | Assign a Call Pickup Group to Directory Numbers, on page 329 | 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. Repeat this procedure for each call pickup group that you create. |
| Step 3 | Configure Partitions for Call Pickup, on page 330 | 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.  
You must complete this procedure for directed call pickup. It is optional for other types of call pickup. |
| Step 4 | Configure Calling Search Spaces | 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 call pickup. It is optional for other types of call pickup. |
<p>| Step 5 | (Optional) Assign a Call Pickup Group to Hunt Pilots, on page 331 | 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. |</p>
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 6** (Optional) Configure notifications:  
  - Configure Call Pickup Notification, on page 331  
  - Configure Call Pickup Notification for a Directory Number, on page 334  
  - Configure BLF Call Pickup Notification, on page 332 | Configure notifications when other members of a pickup group receive a call. You can configure audio or visual notifications, or both. |
| **Step 7** Configure Directed Call Pickup:  
  - Configure Time Period  
  - Configure a Time Schedule  
  - Associate a Time Schedule with a Partition, on page 336 | 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. |
| **Step 8** (Optional) Configure automatic call answering:  
  - Configure Auto Call Pickup, on page 336  
  - Configure BLF Auto Pickup, on page 337 | Enable automatic call answering and configure timers for automatic call answering. |
| **Step 9** Configure phone button templates:  
  - Configure Call Pickup Phone Button Template, on page 338  
  - Associate Call Pickup Button Template with Phone, on page 339  
  - Configure BLF Speed Dial Number for the BLF Call Pickup Initiator, on page 339 | Configure phone button templates for any of the call pickup features that you want to use:  
  - Speed Dial BLF  
  - Call Pickup  
  - Group Call Pickup  
  - Other Group Pickup  
  For Directed Call Pickup, use the Group Call Pickup button. |
| **Step 10** Configure Softkeys for Call Pickup, on page 340  
  - Configure a Sofkey Template for Call Pickup, on page 340 | Configure softkeys for any of the call pickup features that you want to use:  
  - Call Pickup (Pickup)  
  - Group Call Pickup (GPickup) |
### 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 329

**Procedure**

**Step 1** Choose Device > Phone or Call Routing > Directory Number.

**Step 2** Enter the appropriate search criteria to find the phone or directory number that you want to assign to a call pickup group and click Find. A list of phones or directory numbers that match the search criteria displays.

---

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Associate a Softkey Template with a Common Device Configuration, on page 341</td>
<td>• Other Group Pickup (OPickup) For Directed Call Pickup, use the Group Call Pickup softkey.</td>
</tr>
<tr>
<td>• Associate a Softkey Template with a Phone, on page 343</td>
<td></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 329

**Procedure**

**Step 1** Choose Device > Phone or Call Routing > Directory Number.

**Step 2** Enter the appropriate search criteria to find the phone or directory number that you want to assign to a call pickup group and click Find. A list of phones or directory numbers that match the search criteria displays.
Step 3  Choose the phone or directory number to which you want to assign a call pickup group.

Step 4  From the **Association Information** list in the **Phone Configuration** window, choose the directory number to which the call pickup group will be assigned.

Step 5  From the **Call Pickup Group** drop-down list that displays in the Call Forward and Call Pickup Settings area, choose the desired call pickup group.

Step 6  To save the changes in the database, click **Save**.

---

**What to Do Next**

Perform the following tasks:

- Configure Partitions for Call Pickup, on page 330
- Configure Calling Search Spaces

---

**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.

**Procedure**

---

Step 1  In Cisco Unified Communications Manager 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 Related Topics section 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 **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 7** Click Save.

---

**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:

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Hunt Pilot.
**Step 2** 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.
**Step 3** Choose the hunt pilot to which you want to assign a call pickup group.
**Step 4** From the Call Pickup Group drop-down list box that appears in the Hunt Forward Settings area, choose the desired call pickup group.
**Step 5** Click Save.

---

**Configure Call Pickup Notification**

You can configure Call Pickup Notification at the system level, call pickup group level, or individual phone level.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure Call Pickup Notification for a Call Pickup Group, on page 333</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>

| Step 2 Configure Call Pickup Notification for a Directory Number, on page 334 | To configure the type of audio alert to be provided when phone is idle or has an active call. |

| Step 3 Configure BLF Call Pickup Notification, on page 332 | |
Configure BLF Call Pickup Notification

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.

Related Topics

Service Parameter Fields for BLF Call Pickup Notification, on page 332

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 |
| Call Pickup Group Audio Alert Setting of Busy Station | This parameter determines the kind of audio notification that is provided when a phone is busy (in use) and it needs to be alerted regarding an incoming call on its Call Pickup Group. Valid values are as follows:  
  • Disable  
  • Beep Only |
| BLF Pickup Group Audio Alert Setting of Idle Station | This parameter determines the kind of audio notification that is provided when a phone is idle and it needs to be alerted regarding an incoming call on the BLF Pickup Button. Valid values are as follows:  
  • No Ring  
  • Ring Once |
This parameter determines the kind of audio notification that is provided when a phone is busy and it needs to be alerted regarding an incoming call on the BLF Pickup Button. Valid values are as follows:

- No Ring
- Beep Only

### Configure Call Pickup Notification for a Call Pickup Group

**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.

### Related Topics

- [Call Pickup Notification Fields for Call Pickup](#)

### 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 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>
<tr>
<td>Calling Party Information</td>
<td>Check the check box if you want the visual notification message to the call pickup group to include identification of the calling party. The system only makes this setting available when the Call Pickup Group Notification Policy is set to Visual Alert or Audio and Visual Alert.</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.

Procedure

Step 1 From Cisco Unified CM Administration, choose Call Routing > Directory Number. The Find and List Directory Numbers window appears.

Step 2 Enter the search criteria and click Find.

Step 3 Click the directory number for which you want to configure the Call Pickup Notification. The Directory Number Configuration window appears.

Step 4 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.

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.
Configure Directed Call Pickup

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Time Period</td>
<td>Configure time period for members of the associated groups to your group.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure Time Schedule, on page 335</td>
<td>Configure time schedule for members of the associated groups to your group.</td>
</tr>
<tr>
<td><strong>Step 3</strong> Associate a Time Schedule with a Partition, on page 336</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

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified Communications Manager Administration, choose Call Routing &gt; Class of Control &gt; Time Period.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure the fields in the Time Period Configuration window. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

Configure Time Schedule

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Call Routing &gt; Class of Control &gt; Time Schedule.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Configure the fields in the Time Schedule Configuration window. See the online help for more information about the fields and their configuration options.</td>
</tr>
</tbody>
</table>
**Associate a Time Schedule with a Partition**

**Procedure**

**Step 1** Choose Call Routing > Class of Control > Partition.

**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**

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Configure Auto Call Pickup, on page 336</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure BLF Auto Pickup, on page 337</td>
<td></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.
**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 the PickUp, GPickUp, or OPickUp softkey.

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

**Configure BLF Auto Pickup**

**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 values for the following clusterwide service parameters.

- **BLF Pickup Audio Alert Setting of Idle Station**—Select **True** or **False** 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**.

- **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.
Configure Call Pickup Phone 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 Call Pickup Phone Button Template, on page 338</td>
<td>Add Call Pickup feature to the phone button template.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate Call Pickup Button Template with Phone, on page 339</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure BLF Speed Dial Number for the BLF Call Pickup Initiator, on page 339</td>
<td></td>
</tr>
</tbody>
</table>

Configure Call Pickup Phone Button Template

Follow these steps to add Call Pickup feature to the phone button template.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Device Settings > Phone Button Template. The Find and List Phone Button Templates window appears.

Step 2 Click Find. The window displays a list of templates for the supported phones.

Step 3 Perform this step 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 this step if you want to add phone buttons to an existing template.
   a) Enter search criteria and click Find.
   b) Choose an existing template.
   The Phone Button Template Configuration window appears.

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. See the What to Do Next section for more information.
What to Do Next

Associate Call Pickup Button Template with Phone, on page 339

Associate Call Pickup Button Template with Phone

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window is displayed.
Step 2  From the Find and List Phones window, click Find. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.
Step 3  Choose the phone to which you want to add the phone button template. The Phone Configuration window appears.
Step 4  In the Phone Button Template drop-down list, choose the phone button template that contains the new feature button.
Step 5  Click Save. A dialog box is displayed with a message to press Reset to update the phone settings.

Configure BLF Speed Dial Number for the BLF Call Pickup Initiator

Procedure

Step 1  From Cisco Unified CM Administration, choose Device > Phone.
Step 2  Select the phone that you want to use as the BLF call pickup initiator.
Step 3  In the Association pane, Add a new BLF SD link. The Busy Lamp Field Speed Dial Configuration window appears.
Step 4  Select a Directory Number (BLF DN) that should be monitored by the BLF SD button.
Step 5  Check the Call Pickup 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.
Step 6  Click Save.
Configure Softkeys for Call Pickup

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 a Sofkey Template for Call Pickup, on page 340</strong></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>To Associate a Sofkey Template with a Common Device Configuration, on page 341, complete the following subtasks:</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Add a Sofkey Template to Common Device Configuration, on page 342</strong></td>
</tr>
<tr>
<td></td>
<td>• <strong>Associate a Common Device Configuration with a Phone, on page 343</strong></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>Associate a Sofkey Template with a Phone, on page 343</strong></td>
</tr>
</tbody>
</table>

Configure a Sofkey 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>
Procedure

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

**Step 2**
Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.
- d) Click **Save**.

**Step 3**
Perform this step to add softkeys to an existing template.
- a) Enter search criteria and click **Find**.
- b) Choose an 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**
To display the softkey in additional call states, repeat the previous step.

**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. See the What to Do Next section for more information.

---

**What to Do Next**

Perform one of the following tasks:
- **Associate a Softkey Template with a Common Device Configuration**, on page 341
- **Associate a Softkey Template with a Phone**, on page 343

---

**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.

### Procedure

**Step 1**  Add a Softkey Template to Common Device Configuration, on page 342

**Step 2**  Associate a Common Device Configuration with a Phone, on page 343

### Add a Softkey Template to Common Device Configuration

**Procedure**

**Step 1**  From Cisco Unified CM Administration, choose **Device > Device Settings > Common Device Configuration**. The **Find and List Common Device Configuration** window appears.

**Step 2**  Perform this step 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)  In the **Name** field, enter a name for the Common Device Configuration.
   c)  Click **Save**.

**Step 3**  Perform this step to add the softkey template to an existing Common Device Configuration.
   a)  Enter search criteria and click **Find**.
   b)  Choose an existing Common Device Configuration.
      The **Common Device Configuration** window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
Associate a Common Device Configuration with a Phone

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > Phone.
The Find and List Phones window appears.

**Step 2** Find the phone to which 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.
The Find and List Phones window appears.

**Step 2** Choose the phone to which you want to add the softkey template.
The Phone Configuration window appears.

**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 DNs that are configured in Cisco Unified Communications Manager. Use the route plan report to look for overlapping patterns and DNs 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>
</tbody>
</table>
Interaction Feature

Time of Day (TOD) Parameter

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.

Call Accounting

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.

When a call pickup occurs via non-auto call pickup, the system generates one call detail record, which applies to the requesting call that is connected.

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.

Call Forwarding

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.

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>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</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. |
Call Park and Directed Call Park

• Call Park Overview, page 347
• Call Park Prerequisites, page 348
• Call Park Configuration Task Flow, page 348
• Call Park Interactions and Restrictions, page 357
• Troubleshooting Call Park, page 360
• Directed Call Park Overview, page 361
• Directed Call Park Prerequisites, page 361
• Directed Call Park Configuration Task Flow, page 362
• Directed Call Park Interactions and Restrictions, page 365
• Troubleshooting Directed Call Park, page 368

Call Park Overview

The Call Park feature allows you to place a call on hold so that 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.

**Call Park Prerequisites**

If you are using call park across clusters, you must have partitions and calling search spaces configured.

**Table 13: 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 Unified IP Phones 7900 series (except 7921, 7925, 7935, 7936, 7937)</td>
<td>X</td>
<td></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, 7935, 7936, 7937)</td>
<td>X</td>
<td></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**

**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 ClusterWide Call Park, on page 349</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>
</tbody>
</table>
Configure ClusterWide Call Park

**Procedure**

**Step 1** Choose System > Service Parameters.
**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 Partition, on page 350

---

**Configure Partition**

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.

**Procedure**

**Step 1** In Cisco Unified Communications Manager 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 Related Topics section 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 351

**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 range 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.

**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 353

Related Topics

Call Park Configuration Fields, on page 352

Call Park Configuration Fields

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park Number/Range</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> 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.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> 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.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> 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.</td>
</tr>
<tr>
<td>Description</td>
<td>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 (&quot;), percentage sign (%), ampersand (&amp;), or angle brackets (&lt;&gt;).</td>
</tr>
</tbody>
</table>
Field | Description
--- | ---
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.

Cisco Unified Communications Manager | Using the drop-down list, choose the Cisco Unified Communications Manager to which these call park numbers apply.

---

### 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. The Softkey Template Configuration window appears.

**Step 2** Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.
d) Click **Save**.

**Step 3** Perform this step to add softkeys to an existing template.

a) Enter search criteria and click **Find**.
b) Choose an existing template.

The Softkey Template Configuration window appears.

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

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. See the What to Do Next section for more information.

---

**What to Do Next**

*Associate a Softkey Template with a Common Device Configuration*, on page 354

---

**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 187.

**Procedure**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>Add a Softkey Template to a Common Device Configuration</strong>, on page 186</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>Associate a Common Device Configuration with a Phone</strong>, on page 187</td>
</tr>
</tbody>
</table>
Add a Softkey Template to a Common Device Configuration

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; Device Settings &gt; Common Device Configuration</strong>. The Find and List Common Device Configuration window appears.</td>
</tr>
</tbody>
</table>
| Step 2 | Perform this step 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) In the **Name** field, enter a name for the Common Device Configuration.  
  c) Click **Save**. |
| Step 3 | Perform this step to add the softkey template to an existing Common Device Configuration.  
  a) Enter search criteria and click **Find**.  
  b) Choose an existing Common Device Configuration.  
  The Common Device Configuration window appears. |
| 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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.  
  • If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices. |

What to Do Next

Associate a Common Device Configuration with a Phone, on page 187

Associate a Common Device Configuration with a Phone

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
</table>
| Step 1 | From Cisco Unified CM Administration, choose **Device > Phone**.  
  The Find and List Phones window appears. |
| Step 2 | Find the phone to which 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**

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.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Phone</strong>. The <strong>Find and List Phones</strong> window appears.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Choose the phone to which you want to add the softkey template. The <strong>Phone Configuration</strong> window appears.</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 Call Park Button**

**Configure a Phone Button Template for Call Park**

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose <strong>Device &gt; Device Settings &gt; Phone Button Template</strong>. The <strong>Find and List Phone Button Templates</strong> window appears.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Click <strong>Find</strong>. The window displays a list of templates for the supported phones.</td>
</tr>
</tbody>
</table>
| Step 3 | Perform this step 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 this step if you want to add phone buttons to an existing template.  
  a) Enter search criteria and click **Find**.  
  b) Choose an existing template.  
  The **Phone Button Template Configuration** window appears. |
| 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. See the What to Do Next section for more information.

---

**What to Do Next**

*Associate a Button Template with a Phone*, on page 211

---

**Associate a Button Template with a Phone**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window is displayed.

**Step 2** From the **Find and List Phones** window, click **Find**. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.

**Step 3** Choose the phone to which you want to add the phone button template. The **Phone Configuration** window appears.

**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 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>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>-------------------------</td>
<td>-------------------------------------------------------------------------------------------------</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.</td>
</tr>
<tr>
<td></td>
<td>Music On Hold allows two types of hold:</td>
</tr>
<tr>
<td></td>
<td>• User hold—The system invokes this type of hold when a user presses the Hold button or Hold</td>
</tr>
<tr>
<td></td>
<td>softkey.</td>
</tr>
<tr>
<td></td>
<td>• Network hold—This type of hold takes place when a user activates the Transfer, Conference,</td>
</tr>
<tr>
<td></td>
<td>or Call Park feature, and the hold automatically gets invoked.</td>
</tr>
<tr>
<td>Route Plan Report</td>
<td>The route plan report displays the patterns and directory numbers that are configured in Cisco</td>
</tr>
<tr>
<td></td>
<td>Unified Communications Manager. Use the route plan report to look for overlapping patterns and</td>
</tr>
<tr>
<td></td>
<td>directory numbers before assigning a directory number to Call Park.</td>
</tr>
<tr>
<td>Calling Search Space and</td>
<td>Assign the call park directory number or range to a partition to limit call park access to users</td>
</tr>
<tr>
<td>Partitions</td>
<td>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</td>
</tr>
<tr>
<td></td>
<td>B, and user B parks the call. User B retrieves the call and then decides to send the call to</td>
</tr>
<tr>
<td></td>
<td>a voice-messaging mailbox by pressing the iDivert or Divert softkey. User A receives the</td>
</tr>
<tr>
<td></td>
<td>voice mail greeting of user B.</td>
</tr>
<tr>
<td>Barge</td>
<td>• Barge with Call Park—The target phone (the phone that is being barged upon) controls the</td>
</tr>
<tr>
<td></td>
<td>call. The barge initiator “piggybacks” on the target phone. The target phone includes most</td>
</tr>
<tr>
<td></td>
<td>of the common features, even when the target is being barged; therefore, the barge initiator</td>
</tr>
<tr>
<td></td>
<td>has no feature access. When the target parks a call, the barge initiator then must release its</td>
</tr>
<tr>
<td></td>
<td>call (the barge).</td>
</tr>
<tr>
<td></td>
<td>• eBarge with Call Park—The target and barge initiator act as peers. The eBarge feature uses a</td>
</tr>
<tr>
<td></td>
<td>conference bridge, which causes it to function like a MeetMe conference. Both phones (target</td>
</tr>
<tr>
<td></td>
<td>and barge initiator) have full access to their features.</td>
</tr>
<tr>
<td>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Directed Call Park</td>
<td>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.</td>
</tr>
<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 subscriber2 registers first and the line is created in subscriber2 and the publisher node, the line belongs to subscriber2. 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>
</tbody>
</table>
Restriction Feature

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.

Delete Server

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 > Server), the node deletion fails. Before you can delete the node, you must delete the call park numbers in Cisco Unified Communications Manager Administration.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<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 Cluster Wide Call Park, on page 349

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
A user can park and retrieve a call by using directed call park from any phone that can perform a transfer. Cisco VG248 Analog Phone Gateways also support directed call park.

The following IP phones (SCCP and SIP) support directed call park BLF:

- Cisco Unified IP Phones 6900 Series (except 6901 and 6911)
- Cisco Unified IP Phones 7900 Series (except 7906, 7911, 7936, 7937)
- Cisco Unified Wireless IP Phone 7925
- Cisco Unified IP Phone Expansion Module (7914, 7915, 7916)
- Cisco Unified IP Color Key Expansion Module
- Cisco Unified IP Phones 8900 Series
- Cisco Unified IP Phones 9900 Series

The following phones (SCCP) support directed call park BLF
Cisco Unified IP Phones 7940, 7960
## Directed Call Park Configuration Task Flow

<table>
<thead>
<tr>
<th>Procedure</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure ClusterWide Directed Call Park, on page 362</td>
<td>To configure clusterwide parameter for directed call park.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure a Directed Call Park Number, on page 362</td>
<td>To add, copy, and update a single Directed Call Park extension number or range of extension numbers.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure BLF/Directed Call Park Buttons, on page 364</td>
<td>Configure a phone button template for BLF/Directed Call Park.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Synchronize Directed Call Park with Affected Devices</td>
<td>Synchronize Directed Call Park with Affected Devices</td>
</tr>
</tbody>
</table>

### Configure ClusterWide Directed Call Park

**Procedure**

1. In Cisco Unified CM Administration, choose **System > Service Parameters**.
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 362

### 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.
### Procedure

**Step 1** Choose Call Routing &gt; 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 364

### Related Topics

Directed Call Park Configuration Settings, on page 363

---

### 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>
</tbody>
</table>
### Configure BLF/Directed Call Park Buttons

**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.

**Related Topics**

BLF/Directed Call Park Configuration Fields, on page 365
BLF/Directed Call Park Configuration Fields

Table 14: 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>
<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>
</tbody>
</table>

Note: 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.

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>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Music On Hold</td>
<td>Directed Call Park uses the Default Network Hold MOH Audio Source for music on hold.</td>
</tr>
<tr>
<td>Calling Search Space and Partitions</td>
<td>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.</td>
</tr>
</tbody>
</table>
### Directed Call Park Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Immediate Divert</td>
<td>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.</td>
</tr>
</tbody>
</table>
| 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 to 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. 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>
</tbody>
</table>
| Directed Call Park for phones that are running SIP | The following limitations apply to Directed Call Park for phones that are running SIP:  
  - Directed Call Park gets invoked by using the Transfer softkey on Cisco Unified IP Phones 7940 and 7960 that are running SIP.  
  - 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.  
  - 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. |
Troubleshooting Directed Call Park

User Cannot Retrieve Parked Calls

Problem
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".

Solution
Ensure that the user dials the retrieval prefix followed by the directed call park number.

User Cannot Park Calls

Problem
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.

Solution
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

Problem
User cannot park calls. The user receives a reorder tone after the reversion timer expires.

Solution
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

Problem
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.

Solution
Ensure that the dialed number is configured as a directed call park number.

**User Cannot Park a Call at a Number Within The Range**

**Problem**
After configuring a range of directed call park numbers, the user cannot park a call at a number within the range.

**Solution**
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**

**Problem**
Parked calls revert too quickly.

**Solution**
Set the Call Park Reversion Timer to a longer duration.

**Park Slot Unavailable**

**Problem**
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".

**Solution**
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**

**Problem**
Parked calls do not revert to the number that parked the call.

**Solution**
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**

**Problem**
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.

**Solution**
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.
Extension Mobility

- Extension Mobility Overview, page 371
- Extension Mobility Prerequisites, page 371
- Extension Mobility Configuration Task Flow, page 372
- Extension Mobility Interactions and Restrictions, page 380
- Extension Mobility Troubleshooting, page 382

Extension Mobility Overview

Cisco Extension Mobility allows users to temporarily access their Cisco Unified IP Phone settings, such as line appearances, services, and speed dials, from other Cisco Unified IP Phones.

Extension Mobility functions extend to most Cisco Unified IP Phones. You can configure each Cisco Unified IP Phone to support Cisco Extension Mobility by configuring a Device Profile in Cisco Unified Communications Manager Administration.

Extension Mobility Prerequisites

- A TFTP server that is reachable. You can optionally install TFTP and Cisco Unified Communications Manager on the same server.
- Extension mobility functionality extends to most Cisco Unified IP Phones. Check the Cisco Unified IP Phone documentation to verify that Cisco Extension Mobility is supported.
Extension Mobility 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>Generate a report to identify devices that support the Extension Mobility feature.</td>
</tr>
<tr>
<td>Generate a Phone Feature List, on page 7</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure the Extension Mobility Phone Service so that phones can later subscribe to Extension Mobility.</td>
</tr>
<tr>
<td>Activate Extension Mobility Services, on page 372</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure an Extension Mobility Device Profile which 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 of this profile.</td>
</tr>
<tr>
<td>Configure the Cisco Extension Mobility Phone Service, on page 373</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Associate a device profile to a user so that user can access the Extension Mobility function from a phone.</td>
</tr>
<tr>
<td>Create an Extension Mobility Device Profile, on page 374</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Subscribe IP phones and device profiles to the Extension Mobility service so that users can log in, use, and log out of Extension Mobility.</td>
</tr>
<tr>
<td>Associate a Device Profile to a User, on page 374</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>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 Cisco Unified IP Phone with the Change Credential phone service.</td>
</tr>
<tr>
<td>Subscribe to Extension Mobility, on page 375</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Optional. If you want to modify the behavior of Extension Mobility, configure the service parameters.</td>
</tr>
<tr>
<td>Configure the Change Credential IP Phone Service, on page 376</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>Activate Extension Mobility Services</td>
</tr>
<tr>
<td>Configure Extension Mobility Service Parameters, on page 376</td>
<td></td>
</tr>
</tbody>
</table>

Activate Extension Mobility Services

Procedure

| Step 1 | In Cisco Unified Serviceability, choose Tools > Service Activation. |
| Step 2 | From the Server drop-down list, choose a node. |
| Step 3 | Activate the following services: |
| | a) Cisco CallManager |
| | b) Cisco Tftp |
c) Cisco Extension Mobility

Step 4  Click Save.
Step 5  Click OK.

What to Do Next

Configure the Cisco Extension Mobility Phone Service, on page 373

Configure the Cisco Extension Mobility Phone Service

Configure the Extension Mobility Phone Service so that phones can later subscribe to Extension Mobility.

Before You Begin

Activate Extension Mobility Services, on page 372

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 a name for the service.
For Java MIDlet services, the service name must exactly match the name that is defined in the Java Application Descriptor (JAD) file.
Step 4  In the Service URL field, enter the Service URL.
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.
Example:
http://123.45.67.89:8080/emapp/EMAppServlet?device=#DEVICENAME#
Step 5  In the Service Category field, choose whether the service is based on XML or Java MIDlet.
Step 6  In the Service Type field, choose whether the service will be provisioned to the Services, Directories, or Messages button.
Step 7  For Java MIDlet services: in the Service Vendor field, enter the service vendor that exactly matches the vendor that is defined in the Java Application Descriptor (JAD) file. You can leave this field blank for XML services.
Note  A value for Service Version is not required.
Step 8  Click Save.

What to Do Next

Create an Extension Mobility Device Profile, on page 374
Create an Extension Mobility Device Profile

Configure an Extension Mobility Device Profile which 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 of this profile.

**Before You Begin**

*Configure the Cisco Extension Mobility Phone Service, on page 373*

**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; Device Profile.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Perform one of the following tasks:</td>
</tr>
<tr>
<td></td>
<td>• To modify the settings for an existing device profile, enter search criteria, click Find, and choose an existing device profile from the resulting list.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Configure the fields. See the online help for more information about the fields and their configuration options.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click Save. The window refreshes.</td>
</tr>
<tr>
<td>Step 5</td>
<td>From the Association Information section, click Add a new DN.</td>
</tr>
<tr>
<td>Step 6</td>
<td>In the Directory Number field, enter the directory number and click Save. The following prompt appears: Changes to Line or Directory Number settings require restart.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click Reset and follow the prompts.</td>
</tr>
</tbody>
</table>

**What to Do Next**

*Associate a Device Profile to a User, on page 374*

**Associate a Device Profile to a User**

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.

**Before You Begin**

*Create an Extension Mobility Device Profile, on page 374*
**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**, move the device profile from **Available Profiles** to **Controlled Profiles**.

**Step 4**
Click **Save**.
The device profile is associated to the user.

**What to Do Next**

**Subscribe to Extension Mobility**, on page 375

---

**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 374

**Procedure**

**Step 1**
From the Related Links drop-down list on the **Device** or **Device Profile** window, choose **Subscribe/Unsubscribe Services**.
A separate **Subscribed Cisco IP Phone Services** window appears.

**Step 2**
Click **Go**.

**Step 3**
From the **Select a Service** drop-down list, choose the Extension Mobility service.

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

**Step 5**
Click **Subscribe**.
The new service appears under Subscribed Services.

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

**What to Do Next**

**Configure the Change Credential IP Phone Service**, on page 376
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 Cisco Unified IP Phone with the Change Credential phone service.

**Before You Begin**

Subscribe to Extension Mobility, on page 375

**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.

The Change Credential IP phone service appears under Subscribed Services.

**Step 13** Click Save.

**What to Do Next**

(Optional) Configure Extension Mobility Service Parameters, on page 376

Configure Extension Mobility Service Parameters

**Before You Begin**

Activate Extension Mobility Services, on page 372
Procedure

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

Step 2 In the **Server** field, choose the server 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. See the Related Topics section for more information about these service parameters and their configuration options.

Step 5 Click **Save**.

Related Topics

*Extension Mobility Service Parameters, on page 377*

### 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 <strong>True</strong> to specify a maximum time for local logins. After this time, the system automatically logs out the device. <strong>False</strong>, which is the default setting, means that no maximum time for logins exists. To set an automatic logout, you must choose <strong>True</strong> 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>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 if the Enforce Intra-cluster Maximum Login Time parameter is set to <strong>False</strong>. Valid values are between 0:01 and 168:00 in the format HHH:MM, where HHH represents the number of hours and MM represents the number of minutes.</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>
<tr>
<td>Service Parameter</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| Intra-cluster Multiple Login Behavior                  | **Description**: Choose one of the following options:  
  - **Multiple Logins Allowed**—A user can log in to more than one device at a time.  
  - **Multiple Logins Not Allowed**—The second and subsequent login attempts after a user successfully logs in once will fail.  
  - **Auto Logout**—After a user logs in to a second device, the Cisco Unified Communications Manager automatically logs the user out of the first device.  

  **Note**: For EMCC, multiple logins are always allowed. |
| 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 lower case. |
| Remember the Last User Logged In                       | When you choose **False**, the system does not remember the last user who logged into the phone. Use this option when the user will 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 can be 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 userID 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 only 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. |
<table>
<thead>
<tr>
<th>Service Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Validate IP Address</td>
<td>This parameter sets whether validation occurs on the IP address of the source that is requesting login or logout.</td>
</tr>
<tr>
<td></td>
<td>If the parameter is set to <strong>True</strong>, the IP address from which a Cisco Extension Mobility log in or log out request occurs 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.</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>If the parameter is set to <strong>False</strong>, the Cisco Extension Mobility log in or log out request is not validated.</td>
</tr>
<tr>
<td></td>
<td>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 in conjunction with log ins from separate trusted proxy servers for remote devices.</td>
</tr>
<tr>
<td>Trusted List of IPs</td>
<td>This parameter appears as a text box (the maximum length is 1024 characters). You can enter strings of trusted IP addresses or hostnames, separated by semicolons, in the text box. IP address ranges and regular expressions are not supported.</td>
</tr>
<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 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 you enter takes effect only if the Validate IP Address parameter is <strong>True</strong>.</td>
</tr>
</tbody>
</table>
Extension Mobility Interactions and Restrictions

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>
</tbody>
</table>
| BLF Presence         | When you configure BLF/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/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. |
| Call Display Restrictions | When you enable Call Display Restrictions with Cisco Extension Mobility, 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 Ignore Presentation Indicators (internal calls only) check box in both the Device Profile Configuration window and the Phone Configuration window. |
| 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. |
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 intercome 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 IPv4, so you cannot use phones with an IP Addressing Mode of IPv6 Only for Cisco Extension Mobility. If you want to use Cisco Extension Mobility with the phone, you must configure the phone with an IP Addressing Mode of IPv4 Only or 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.

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 Extenion 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>
</tbody>
</table>
Extension Mobility Troubleshooting

**Troubleshoot Extension Mobility**

- Configure the Cisco Extension Mobility trace directory and enable debug tracing by performing the following steps:
  a) From Cisco Unified Serviceability, select **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.

---

**Feature** | **Restriction**
---|---
**IP Phones** | 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.

**Locale** | 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.

**Log Out** | 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 Communications Manager Administration.

**Secure Tone** | Cisco Extension Mobility and Join Across Line services are disabled on protected phones.

**User Group** | Although you can add users to the Standard EM Authentication Proxy Rights user group, those users are not authorized to authenticate by proxy.
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.

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 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 373

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 Cisco Unified Serviceability > Tools > Control Center—Feature.

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 Cisco Unified Serviceability > Tools > Control Center—Network Services.

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 Cisco Unified Serviceability > Tools > Control Center—Network Services.
- If you get this error when you select Extension Mobility service, check that the Cisco Extension Mobility Application service is running by selecting Cisco Unified Serviceability > Tools > Control Center—Network Services.

Phone Resets

Problem After users log in or log out, their phones reset instead of restarting.

Possible Cause Locale change is the probably 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 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 Cisco Extension Mobility Cross Cluster (EMCC) feature allows a user of one Cisco Unified Communications Manager cluster (the home cluster) to log in to a Cisco Unified IP Phone of another Cisco Unified Communications Manager cluster (the visiting cluster) during travel, as if the user is using the IP phone at the home office.

Extension Mobility Cross Cluster Prerequisites

- Cisco Extension Mobility service
- Other call-control entities that support and use the Cisco Extension Mobility Cross Cluster 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
- Supported phones in secure or nonsecure mode
## Extension Mobility Cross Cluster 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 7</td>
<td>Generate a report to identify devices that support the Extension Mobility Cross Cluster feature.</td>
</tr>
<tr>
<td>Step 2</td>
<td>To Configure Extension Mobility, on page 389, perform the following subtasks:</td>
<td>Cisco Extension Mobility allows users to temporarily access their Cisco Unified IP Phone settings, such as line appearances, services, and speed dials, from other Cisco Unified IP Phones. Perform this task flow on both home and remote clusters.</td>
</tr>
<tr>
<td></td>
<td>• Activate Services for EMCC, on page 389</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Create Phone Service, on page 390</td>
<td></td>
</tr>
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<td></td>
<td>• Configure an EMCC Device Profile, on page 391</td>
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</tr>
<tr>
<td></td>
<td>• Enable Extension Mobility Cross Cluster for a User, on page 391</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Subscribe Devices to Extension Mobility, on page 392</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>To Configure Certificates, on page 393, 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></td>
<td>• Activate the Bulk Provisioning Service, on page 393</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure Bulk Certificate Management</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure Bulk Certificate Management and Export Certificates, on page 393</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Consolidate the Certificates, on page 394</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Import the Certificates, on page 395</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>To Add Extension Mobility Cross Cluster Devices, on page 395, perform the following subtasks:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Create Common Device Configuration, on page 395</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Configure an EMCC Template, on page 396</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Set the Default Template, on page 396</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Insert EMCC Devices, on page 397</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>Configure a Geolocation Filter, on page 397</td>
<td></td>
</tr>
</tbody>
</table>
### Purpose

Select values for the feature parameters that you configured, such as the geolocation filter.

Configure trunks to process inbound or outbound traffic for intercluster PSTN access and RSVP Agent services.

Configure EMCC Intercluster Service Profile, on page 402

Configure Remote Cluster Services, on page 403

---

### Configure Extension Mobility

Perform these steps on the home and remote clusters.

#### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Activate Services for EMCC, on page 389 Activate Extension Mobility Services in Cisco Unified Serviceability.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Create Phone Service, on page 390 Create the Extension Mobility Phone Service.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure an EMCC Device Profile, on page 391 Create a device profile to map settings onto a real device when a user logs into Extension Mobility Cross Cluster.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Enable Extension Mobility Cross Cluster for a User, on page 391 Enable Extension Mobility Cross Cluster for the user.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Subscribe Devices to Extension Mobility, on page 392 Enable Extension Mobility on devices and subscribe to the service if you have not set up an enterprise subscription for all devices.</td>
</tr>
</tbody>
</table>

---

### Activate Services for EMCC

#### Procedure

1. In Cisco Unified Serviceability, choose **Tools > Service Activation**.
2. From the **Server** drop-down list, choose a node.
3. Activate the following services:
a) Cisco CallManager
b) Cisco Tftp
c) Cisco Extension Mobility

**Step 4** Click Save.
**Step 5** Click OK.

---

**What to Do Next**

Create Phone Service, on page 390

---

**Create Phone Service**

**Before You Begin**

Activate Services for EMCC, on page 389

**Procedure**

**Step 1** In Cisco Unified CM Administration, choose Device > Device Settings > Phone Services.

**Step 2** Click Add New.

**Step 3** In the Service Name field, enter a name for the service.
For example, enter a name such as Extension Mobility or EM. For Java MIDlet services, the service name must exactly match the name that is defined in the Java Application Descriptor (JAD) file.

**Step 4** In the Service URL field, enter the service URL in the following format:
http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#.

**Step 5** (Optional) If you want to create a secure URL using HTTPS, enter the same URL in the Secure-Service URL Field.
Change http:// to https://.

**Step 6** Use the default values for the Service Category and Service Type fields.

**Step 7** Check the Enable check box.

**Step 8** (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 9** Click Save.

---

**What to Do Next**

Configure an EMCC Device Profile, on page 391
Configure an EMCC Device Profile

The device profile is used to map onto a physical device when a user logs in to Extension Mobility Cross Cluster.

Before You Begin

Create Phone Service, on page 390

Procedure

Step 1 In Cisco Unified Communications Manager 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 online help for more information about the fields and their configuration options.

Step 4 Click Save. The page refreshes.

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 391

Enable Extension Mobility Cross Cluster for a User

Before You Begin

Configure an EMCC Device Profile, on page 391

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select 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** Select the device profile from the **Available Profiles** list pane in the **Extension Mobility** pane.

**Step 5** Use the Down arrow to move the device profile to the **Controlled Profiles** list pane.

**Step 6** Select **Save**.

---

**What to Do Next**

*Subscribe Devices to Extension Mobility, on page 392*

---

**Subscribe Devices to Extension Mobility**

Subscribe devices to Extension Mobility if you have not enabled an enterprise subscription service.

**Before You Begin**

*Enable Extension Mobility Cross Cluster for a User, on page 391*

---

**Procedure**

**Step 1** In 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** Click **Save**, and then close the popup window.

**Step 10** In the **Phone Configuration** window, click **Save**.

**Step 11** Click **OK** if prompted.
## Configure Certificates

**Procedure**

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<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
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<tbody>
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<td>Step 1</td>
<td>Activate the Bulk Provisioning Service, on page 393</td>
<td>Activate the Bulk Provisioning Service in Cisco Unified Serviceability.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure Bulk Certificate Management and Export Certificates, on page 393</td>
<td>Configure Bulk Certificate Management in Cisco Unified OS Administration to export the certificates from both the home and remote clusters.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Consolidate the Certificates, on page 394</td>
<td>Consolidate the certificates. You can perform this step on either cluster.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Import the Certificates, on page 395</td>
<td>Import the certificates back into the home and remote clusters.</td>
</tr>
</tbody>
</table>

### Activate the Bulk Provisioning Service

**Procedure**

1. In Cisco Unified Serviceability, select **Tools > Service Activation**.
2. In 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 393**

### Configure Bulk Certificate Management and Export Certificates

This procedure creates a PKCS12 file that contains certificates for all nodes in the cluster. Perform this procedure on both the home and remote clusters.

#### 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, TFTP, or CAPF certificates are regenerated on any of the cluster nodes.
**Before You Begin**

*Activate the Bulk Provisioning Service, on page 393*

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified Operating System Administration, choose <strong>Security &gt; Bulk Certificate Management</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>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.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click <strong>Save</strong>.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Click <strong>Export</strong>.</td>
</tr>
<tr>
<td>Step 5</td>
<td>In the <strong>Bulk Certificate Export</strong> window, choose <strong>All</strong> for the <strong>Certificate Type</strong> field.</td>
</tr>
<tr>
<td>Step 6</td>
<td>Click <strong>Export</strong>.</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click <strong>Close</strong>.</td>
</tr>
</tbody>
</table>

**What to Do Next**

*Consolidate the Certificates, on page 394*

**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**

- Only one of the participating clusters needs to perform consolidation.
- 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 393*

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In Cisco Unified Operating System Administration, choose <strong>Security &gt; Bulk Certificate Management &gt; Consolidate &gt; Bulk Certificate Consolidate</strong>.</td>
</tr>
<tr>
<td>Step 2</td>
<td>In the <strong>Certificate Type</strong> field, choose <strong>All</strong>.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click <strong>Consolidate</strong>.</td>
</tr>
</tbody>
</table>

**What to Do Next**

*Import the Certificates, on page 395*
Import the Certificates

Import the certificates onto the home and remote clusters.

Note
After an upgrade, these certificates are preserved. You do not need to reimport or reconsolidate certificates.

Before You Begin
Consolidate the Certificates, on page 394

Procedure

Step 1 In Cisco Unified Operating System Administration, choose Security > Bulk Certificate Management > Import > Bulk Certificate Import.
Step 2 From the Certificate Type drop-down list, choose All.
Step 3 Choose Import.

Add Extension Mobility Cross Cluster Devices

Procedure

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<td></td>
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<tr>
<td>Step 2 Configure an EMCC Template, on page 396</td>
<td></td>
</tr>
<tr>
<td>Step 3 Set the Default Template, on page 396</td>
<td>Set the EMCC template that you created as the default template.</td>
</tr>
<tr>
<td>Step 4 Insert EMCC Devices, on page 397</td>
<td></td>
</tr>
</tbody>
</table>

Create Common Device Configuration

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose Device > Device Settings > Common Device Configuration.
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 EMCC Template, on page 396

Configure an EMCC Template

Before You Begin
Create Common Device Configuration, on page 395

Procedure

Step 1 In Cisco Unified CM Administration, choose Bulk Administration > EMCC > EMCC Template.
Step 2 Click Add New.
Step 3 Configure the fields on the EMCC Template Configuration window. See the online help for more information about the fields and their configuration options.
Step 4 Click Save.

What to Do Next
Set the Default Template, on page 396

Set the Default Template

Before You Begin
Configure an EMCC Template, on page 396

Procedure

Step 1 In Cisco Unified CM Administration, choose Bulk Administration > EMCC > Insert/Update EMCC.
Step 2 Click Update EMCC Devices.
Step 3 From the Default EMCC Template drop-down list, choose the EMCC Device Template that you configured.
Step 4 Click Run Immediately.
Step 5 Click Submit.
Step 6 Verify the success of the job:
   a) Choose Bulk Administration > Job Scheduler.
b) Locate the Job ID of your job.

What to Do Next
Insert EMCC Devices, on page 397

Insert EMCC Devices

Before You Begin
Set the Default Template, on page 396

Procedure

Step 1 In Cisco Unified CM Administration, choose Bulk Administration > EMCC > Insert/Update EMCC.
Step 2 Click Insert EMCC Devices.
Step 3 Enter the number of EMCC devices you are adding in the Number of EMCC Devices to be added field.
Step 4 Click Run Immediately and click Submit.
Step 5 Refresh this window and verify that the Number of EMCC Devices already in database value shows the number of devices that you added.

Configure a Geolocation Filter

Procedure

Step 1 In Cisco Unified CM Administration, choose System > Geolocation Filter.
Step 2 Click Add New.
Step 3 Configure the fields on the Geolocation Filter 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 EMCC Feature Parameters, on page 397

Configure EMCC Feature Parameters

Before You Begin
- Configure Bulk Certificate Management
- Configure EMCC Feature Parameters, on page 397
Procedure

Step 1  In Cisco Unified CM Administration, choose Advanced Features > EMCC > EMCC Feature Configuration.

Step 2  Configure the fields on the EMCC Feature Configuration window. See the Related Topics section for more information about the fields and their configuration options.

Step 3  Click Save.

What to Do Next

Configure Intercluster SIP Trunk, on page 401

Related Topics

   EMCC Feature Parameter Configuration Fields, on page 398

EMCC Feature Parameter Configuration Fields

<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 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>
</tbody>
</table>
| Default Interval for Expired EMCC Device Maintenance | 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. |
<table>
<thead>
<tr>
<th><strong>EMCC Parameter</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<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. |
<p>| EMCC Geolocation Filter                                                            | Choose the geolocation filter that you have configured for use with the Cisco Extension Mobility Cross Cluster feature. 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 gets applied. |</p>
<table>
<thead>
<tr>
<th><strong>EMCC Parameter</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>EMCC Region Max Audio Bit Rate</td>
<td>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 EMCC Region Max Audio Bit Rate.</td>
</tr>
<tr>
<td>EMCC Region Max Video Call Bit Rate</td>
<td>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. Note: All participating EMCC clusters must specify the same EMCC Region Max Video Call Bit Rate.</td>
</tr>
<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. Note: To allow two-way audio on EMCC calls, all participating EMCC clusters must use the same EMCC Region Link Loss Type. 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. Valid values are the following: • Lossy (a link where some packet loss can or may occur, for example, DSL) • Low Loss (a link where low packet loss occurs, for example, T1). When you set this parameter to Lossy, 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. When this parameter is set to Low Loss, 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. The only difference in the audio codec preference ordering between the Low Loss and Lossy options is that G.722 is preferred over iSAC (Internet Speech Audio Codec) 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. The default value is Low Loss.</td>
</tr>
</tbody>
</table>
### EMCC Parameter

<table>
<thead>
<tr>
<th>EMCC Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<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. 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. 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 Cisco Unified Communications Manager 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 Cisco Unified Communications Manager 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>
<tr>
<td>Remote Cluster Update Interval</td>
<td>Specify an interval, in minutes, during which the Cisco Extension Mobility service on the local Cisco Unified Communications Manager 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.</td>
</tr>
</tbody>
</table>

### Configure Intercluster SIP Trunk

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 EMCC SIP trunks.

**Before You Begin**

Configure EMCC Feature Parameters, on page 397
Procedure

Step 1 In Cisco Unified Communications Manager 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 EMCC Intercluster Service Profile, on page 402

Configure EMCC Intercluster Service Profile

Before You Begin
Configure Intercluster SIP Trunk, on page 401

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose Advance Features > EMCC > EMCC Intercluster Service Profile.
Step 2 Check the Active check box in the EMCC pane.
Step 3 Check the Active check box in the PSTN Access pane.
Step 4 From the PSTN Access SIP Trunk drop-down list, choose a SIP trunk that you configured.
Step 5 Check the Active check box in the RSVP Agent pane.
Step 6 From the RSVP Agent SIP Trunk drop-down list, choose another SIP trunk that you configured.
Note If you configured only one trunk, you can choose the same trunk for the RSVP Agent SIP Trunk and the PSTN Access SIP Trunk.
Step 7 Click Validate to validate your changes.
Step 8 If no failure messages appear in the popup window, click Save.

What to Do Next
Configure Remote Cluster Services, on page 403
Configure Remote Cluster Services

Before You Begin
Configure EMCC Intercluster Service Profile, on page 402

Procedure

**Step 1** From Cisco Unified CM Administration, choose Advanced Features > Cluster View.

**Step 2** Click Find to show a list of known remote clusters.

**Step 3** Perform one of the following steps:

- If the remote cluster that you want to configure appears, click the remote cluster name and verify the fields.
- If the remote cluster that you want to configure does not appear, click Add New and configure the following fields:
  1. For the Cluster Id field, ensure that the ID matches the enterprise parameter value of the cluster ID of the other clusters.
  2. In the Fully Qualified Name field, enter the IP address of the remote cluster or a domain name that can resolve to any node on the remote cluster.
  3. Click Save.

**Note** For Extension Mobility Cross Cluster, the TFTP check box should always be disabled.

---

Extension Mobility Cross Cluster Interactions and Restrictions

**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.</td>
</tr>
<tr>
<td></td>
<td>• The system cannot apply Cisco Unified Communications Manager locations and regions across the cluster boundaries.</td>
</tr>
<tr>
<td></td>
<td>• RSVP agent-based CAC uses RSVP agents in the visiting cluster.</td>
</tr>
<tr>
<td>Call Forwarding</td>
<td>Extension Mobility Cross Cluster supports Call Forwarding.</td>
</tr>
</tbody>
</table>
Interaction

**Feature** | **Interaction**
--- | ---
Cisco Extension Mobility login and logout | User authentication takes place across clusters.

Media resources for the visiting phone | 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).

PSTN access for the visiting phone | • E911 calls are routed to the local gateways of the PSTN.
• Local calls are routed to the local gateways of the PSTN.
• Calls terminating to local route groups route to local gateways in the visiting cluster.

Other call features and services | Example restriction: Intercom configuration specifies configuration to a static device, so Cisco Extension Mobility Cross Cluster does not support the Intercom feature.

Security | • Cross-cluster security is provided by default.
• Cisco Unified IP Phones with secure and nonsecure phone security profiles are supported.

Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
</table>
| Unsupported Features | • EMCC does not support the Intercom feature, because Intercom configuration requires a static device.
• Location CAC is not supported, but RSVP-based CAC is supported. |
| EMCC Device Cannot Be Provisioned in More Than One Cluster | 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. |
| Number of EMCC Devices | 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 |
| **Note** | EMCC login does not affect the number of licenses that are used in the home cluster. |
### Visiting Device Logout Limitations

- 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.

- In the visiting cluster, the Phone Configuration 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 due to a reset or a logout from the home cluster.

<table>
<thead>
<tr>
<th>Visiting Device Login Limitations</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>- The Cisco 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 Cisco 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 &quot;Login is unavailable&quot; error message. At this point, a login attempt to EMCC from a visiting phone can fail; the phone displays the &quot;Login is unavailable&quot; 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 Cisco 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 &gt; EMCC &gt; 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.</td>
<td></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 &quot;EMCC&quot; 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 EMApp 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)</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)</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)</td>
<td>EMService Connection Error</td>
<td>The visiting EMApp cannot connect to any Visiting EMService. (The service is down or not activated.) The visiting EMService cannot connect to the Home EMService (the WAN is down or certificates are not trusted.)</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 EMApp. 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>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>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>

### 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) Login is unavailable(0)</td>
<td>Unknown Error</td>
<td>The EMService failed for an unknown reason.</td>
</tr>
<tr>
<td>1</td>
<td>Login is unavailable(1) Login is unavailable(1)</td>
<td>Error on parsing</td>
<td>When the EMService cannot parse the XML request from the EMAppl or EMService. This error occurs when third-party applications send an incorrect query o 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) Login 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) Login 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) Login 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>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>6</td>
<td>Login 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>8</td>
<td>Login 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>
</tbody>
</table>
| 9          | Login is unavailable(9)            | Dir. User Info Error      | 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. |
| 10         | Login is unavailable(10)           | User lacks app proxy rights | The user tries to log in on behalf of another user. By default, a CCMSysUser has administrative rights.                               |
| 11         | Login is unavailable(11)           | Device Does not exist     | The phone record entry is absent in the device table.                                                                                   |
| 12         | Phone record entry is absent in the device table | Dev. Profile not found | No device profile is associated with the remote user.                                                                                  |
| 18         | Login is unavailable(18)           | Another user logged in    | Another user is already logged in on the phone.                                                                                       |
| 19         | Logout is unavailable(19)          | No user logged in         | 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). |

DatabaseErrorLogin is unavailable(6)
Logout is unavailable(6)

Login is unavailable(8)
Logout is unavailable(8)

Login is unavailable(9)
Logout is unavailable(9)

Login is unavailable(10)
Logout is unavailable(10)

Login is unavailable(11)
Logout is unavailable(11)

Phone record entry is absent in the device table

Login is unavailable(18)

Logout is unavailable(19)
<table>
<thead>
<tr>
<th>Error Code</th>
<th>Phone Display</th>
<th>Quick Description</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<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>21</td>
<td>Login is unavailable(21) Logout 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>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) Logout 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>25</td>
<td>Login is unavailable(25)</td>
<td>User logged in elsewhere</td>
<td>The user is currently logged in on some other phone.</td>
</tr>
<tr>
<td>26</td>
<td>Login is unavailable(26) Logout 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>28</td>
<td>Login is unavailable(28) Logout is unavailable(28)</td>
<td>Untrusted IP Error</td>
<td>Occurs when the Validate IP Address 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>29</td>
<td>Login is unavailable(29) Logout 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>30</td>
<td>Login is unavailable(30) Logout is unavailable(30)</td>
<td>Proxy not allowed</td>
<td>When login and logout occur through proxy (&quot;Via&quot; is set in HTTP header) and the Allow Proxy service parameter is set to False.</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>31</td>
<td>Login is unavailable(31)</td>
<td>EMCC Not Activated for the user</td>
<td>Occurs when the <strong>Enable Extension Mobility Cross Cluster</strong> check box is not checked in the <strong>End User Configuration</strong> 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 <strong>Enable Extension Mobility Cross Cluster</strong> check box is not checked in the home cluster).</td>
</tr>
<tr>
<td>42</td>
<td>Login is unavailable(42)</td>
<td>Invalid ClusterID</td>
<td>Occurs when the remote cluster ID is not valid. This error can occur during a remote cluster update.</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>43</td>
<td>Login is unavailable(43)</td>
<td>Device Security mode error</td>
<td>The Device Security Profile that is associated to the EMCC device should be set to Nonsecure for its Device Security Mode.</td>
</tr>
<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>
</tbody>
</table>
CHAPTER 31

Hold Reversion

This chapter provides information about the hold reversion feature which alerts a phone user when a held call exceeds a configured time limit.

- Hold Reversion Overview, page 413
- Hold Reversion Prerequisites, page 414
- Hold Reversion Configuration Task Flow, page 414
- Hold Reversion Interactions and Restrictions, page 417

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
- Cisco Database Layer Monitor service must be running on the same node as the Cisco CallManager service
- Cisco RIS Data Collector 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 must be installed, if you want to use non-English phone locales or country-specific tones

## 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.

### Procedure

<table>
<thead>
<tr>
<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 7</td>
</tr>
<tr>
<td></td>
<td>Run a phone feature list report to determine which phones support the Hold Reversion feature.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Call Focus Priority for Hold Reversion, on page 415</td>
</tr>
<tr>
<td></td>
<td>Configure the call focus priority setting against the device pool for your phones.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Perform one of the following procedures:</td>
</tr>
<tr>
<td></td>
<td>• Configure Hold Reversion Timer Defaults for Cluster, on page 415</td>
</tr>
<tr>
<td></td>
<td>• Configure Hold Reversion Timer Settings for Phone, on page 416</td>
</tr>
<tr>
<td></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>The settings on an individual phone line override the clusterwide service parameter settings.</td>
</tr>
</tbody>
</table>
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.

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 415
- **Configure Hold Reversion Timer Settings for Phone**, on page 416

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 415
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 416

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 415 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>
<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 Park 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.

Shared Lines

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.

If a shared line device disables the feature, hold reversion gets disabled on all other devices that share the line.

Ring Settings

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.

Hold Reversion Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Restriction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Extension Mobility and Cisco Web Dialer</td>
<td>Cisco Extension Mobility and Cisco Web Dialer features do not support the hold reversion feature.</td>
</tr>
<tr>
<td>SCCP phones</td>
<td>This feature does not support SCCP analog phone types, such as ATA 186, DPA-7610, and DPA-7630. Only certain on-net phone devices that are running SCCP on a node can invoke the hold reversion feature.</td>
</tr>
<tr>
<td>Directory numbers</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>Feature</td>
<td>Restriction</td>
</tr>
<tr>
<td>-------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Shared lines</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. If a shared-line device disables this feature, hold reversion gets disabled on all other devices that share this line.</td>
</tr>
<tr>
<td>Ring settings</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>Maximum number of reverted calls</td>
<td>The maximum number of reverted calls on a line equals the maximum number of calls on your system.</td>
</tr>
<tr>
<td>CTI Applications</td>
<td>To enable this feature with CTI applications, ensure that the CTI application is certified to work with this feature and this release. Otherwise, the CTI application may fail because the hold reversion feature may affect existing CTI applications. This feature gets disabled by default. See the following API documents for information about CTI requirements:</td>
</tr>
<tr>
<td></td>
<td>- Cisco Unified TAPI Developers Guide for Cisco Unified Communications Manager</td>
</tr>
<tr>
<td></td>
<td>- Cisco Unified JTAPI Developers Guide for Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Cisco Unified IP Phones</td>
<td>You cannot configure hold reversion settings for DN s 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. When Hold Reversion is configured for the system, the phone must support the feature or the feature does not activate. See the Cisco Unified IP Phone administration guides for Cisco Unified IP Phone models that support hold reversion and this version of Cisco Unified Communications Manager for any phone restrictions with hold reversion.</td>
</tr>
</tbody>
</table>
Hold Reversion Interactions and Restrictions
Chapter 32

Accessing Hunt Groups

• Hunt Group Overview, page 421
• Hunt Group Prerequisites, page 422
• Hunt Group Configuration Task Flow, page 422
• Hunt Group Interactions and Restrictions, page 427

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**

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<th>Command or Action</th>
<th>Purpose</th>
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<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure a Softkey Template for Hunt Group, on page 422</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 424, complete the following subtasks:  
  • Add a Softkey Template to a Common Device Configuration, on page 424  
  • Associate a Common Device Configuration with a Phone, on page 425 | 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 425 | 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 426 | 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. The Softkey Template Configuration window appears.

Step 2
Perform this step 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) In the Softkey Template Name field, enter a new name for the template.
   d) Click Save.

Step 3
Perform this step to add softkeys to an existing template.
   a) Enter search criteria and click Find.
   b) Choose an existing template.

The Softkey Template Configuration window appears.

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

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. See the What to Do Next section for more information.
What to Do Next

Perform one of the following procedures:

- Add a Softkey Template to a Common Device Configuration, on page 424
- Associate a Softkey Template with a Phone, on page 425

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 425.

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 424</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate a Common Device Configuration with a Phone, on page 425</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. The Find and List Common Device Configuration window appears.

**Step 2**  Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
c) Click Save.

**Step 3**  Perform this step to add the softkey template to an existing Common Device Configuration.

a) Enter search criteria and click Find.
b) Choose an existing Common Device Configuration.
The **Common Device Configuration** window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.

---

**What to Do Next**

**Associate a Common Device Configuration with a Phone, on page 425**

---

**Associate a Common Device Configuration with a Phone**

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**
Find the phone to which 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.

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**
Choose the phone to which you want to add the softkey template.
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 Logged Into Hunt Group check box.

b) To log in the phone to the hunt group, ensure that the Logged Into Hunt Group check box is checked.
   Note: The Logged 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 426

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

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>
### Interaction

**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.

### Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multiple Line Groups</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 DNs that belong to multiple line groups, pressing the HLog softkey logs the phone out of all associated line groups.</td>
</tr>
<tr>
<td>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>-------------</td>
</tr>
</tbody>
</table>
| 7940, 7960, and third-party SIP phones | • When a phone that is running SIP (7906, 7911, 7941, 7961, 7970, and 7971) 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 sofkey 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. |
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** service 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>
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<tr>
<th>Command or Action</th>
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</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>Configure Consult and Blind Transfer, on page 432</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 Trnsfer as a softkey and/or button.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>Configure Transfer On-Hook, on page 437</td>
<td>(Optional) Transfer On-Hook is an option to complete call transfers. Press Transfer, 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></td>
<td></td>
</tr>
<tr>
<td>Configure Direct Transfer, on page 437</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>
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</tr>
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<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>Configure a Softkey Template for Transfer, on page 432</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>Configure Transfer Button, on page 435</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. The Softkey Template Configuration window appears.

**Step 2** Perform this step 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) In the Softkey Template Name field, enter a new name for the template.
   d) Click Save.

**Step 3** Perform this step to add softkeys to an existing template.
   a) Enter search criteria and click Find.
   b) Choose an existing template.
   The Softkey Template Configuration window appears.

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

**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. See the What to Do Next section for more information.

**What to Do Next**

Perform one of the following procedures:
   - Associate Transfer Softkey Template with a Common Device Configuration, on page 433
   - Associate Transfer Softkey Template with a Phone, on page 435

**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 435.

**Procedure**

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<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add Transfer Softkey Template to the Common Device Configuration, on page 434</td>
</tr>
</tbody>
</table>

Perform this step to add the transfer softkey template to the Common Device Configuration.

| **Step 2** | Associate a Common Device Configuration with a Phone, on page 435 |

Perform this step to link the transfer softkey Common Device Configuration to a phone.

**Add Transfer Softkey Template to the Common Device Configuration**

**Procedure**

**Step 1**

From Cisco Unified CM Administration, choose **Device > Device Settings > Common Device Configuration**. The **Find and List Common Device Configuration** window appears.

**Step 2**

Perform this step 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) In the **Name** field, enter a name for the Common Device Configuration.

c) Click **Save**.

**Step 3**

Perform this step to add the softkey template to an existing Common Device Configuration.

a) Enter search criteria and click **Find**.

b) Choose an existing Common Device Configuration.

The **Common Device Configuration** window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
What to Do Next

Associate a Common Device Configuration with a Phone, on page 435

Associate a Common Device Configuration with a Phone

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.
Step 2 Find the phone to which 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 Transfer 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.

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window appears.
Step 2 Choose the phone to which you want to add the softkey template. The Phone Configuration window appears.
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 Transfer Button

The procedures in this section describe how to configure the Transfer button.

Procedure

<table>
<thead>
<tr>
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</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Configure a Phone Button Template for Transfer, on page 436</td>
<td>Perform this step to assign Transfer button features to line or speed dial keys.</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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong></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. The Find and List Phone Button Templates window appears.

**Step 2** Click Find.
The window displays a list of templates for the supported phones.

**Step 3** Perform this step 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 this step if you want to add phone buttons to an existing template.
- a) Enter search criteria and click Find.
- b) Choose an existing template.
The Phone Button Template Configuration window appears.

**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. See the What to Do Next section for more information.

**What to Do Next**
Associate Transfer Button Template with a Phone, on page 436

**Associate Transfer Button Template with a Phone**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > Phone.
The Find and List Phones window is displayed.

**Step 2** From the Find and List Phones window, click Find.
A list of phones that are configured on the Cisco Unified Communications Manager is displayed.

**Step 3**

Choose the phone to which you want to add the phone button template. The **Phone Configuration** window appears.

**Step 4**

In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

**Step 5**

Click **Save**. A dialog box is displayed with a message to press **Reset** to update the phone settings.

---

### Configure Transfer On-Hook

**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>From Cisco Unified CM Administration, choose <strong>System &gt; Service Parameters</strong>. The <strong>Service Parameter Configuration</strong> window is displayed.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>From the <strong>Server</strong> drop-down list, choose the server on which you want to configure the parameter.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>From the <strong>Service</strong> drop-down list, choose the <strong>Cisco CallManager (Active)</strong> service.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>In the <strong>Clusterwide Parameters (Device - Phone)</strong>, choose <strong>True</strong> for the <strong>Transfer On-Hook Enabled</strong> service parameter.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click <strong>Save</strong>.</td>
<td></td>
</tr>
</tbody>
</table>

### Configure Direct 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></td>
<td>Configure a Softkey Template for Direct Transfer, on page 437</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Direct Transfer Button, on page 440</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**. The **Softkey Template Configuration** window appears.

**Step 2** Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.
   d) Click **Save**.

**Step 3** Perform this step to add softkeys to an existing template.
   a) Enter search criteria and click **Find**.
   b) Choose an existing template.
   The **Softkey Template Configuration** window appears.

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

**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. See the What to Do Next section for more information.

---

**What to Do Next**

Perform one of the following procedures:

• **Associate Direct Transfer Softkey Template with a Common Device Configuration**, on page 439
• **Associate Direct Transfer Softkey Template with a Phone**, on page 440
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 440

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Add Direct Transfer Softkey Template to the Common Device Configuration, on page 439</td>
<td>Perform this step to add Direct Transfer softkey template to the Common Device Configuration.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Associate a Common Device Configuration with a Phone, on page 440</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. The Find and List Common Device Configuration window appears.

Step 2 Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
c) Click Save.

Step 3 Perform this step to add the softkey template to an existing Common Device Configuration.

a) Enter search criteria and click Find.
b) Choose an existing Common Device Configuration.
The Common Device Configuration window appears.

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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.
If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.

What to Do Next
Associate a Common Device Configuration with a Phone, on page 440

Associate a Common Device Configuration with a Phone

Procedure

Step 1
From Cisco Unified CM Administration, choose Device > Phone.
The Find and List Phones window appears.

Step 2
Find the phone to which 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

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.

Procedure

Step 1
From Cisco Unified CM Administration, choose Device > Phone.
The Find and List Phones window appears.

Step 2
Choose the phone to which you want to add the softkey template.
The Phone Configuration window appears.

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 Direct Transfer Button

The procedures in this section describe how to configure the Direct Transfer button.
## 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 Phone Button Template for Direct Transfer</strong>, on page 441</td>
<td>Perform this step to assign Direct Transfer button features to line or speed dial keys.</td>
</tr>
<tr>
<td>Step 2</td>
<td><strong>Associate Direct Transfer Button Template with a Phone</strong>, on page 442</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**

#### Step 1
From Cisco Unified CM Administration, choose **Device > Device Settings > Phone Button Template**. The **Find and List Phone Button Templates** window appears.

#### Step 2
Click **Find**. The window displays a list of templates for the supported phones.

#### Step 3
Perform this step 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 this step if you want to add phone buttons to an existing template.  
   a) Enter search criteria and click **Find**.  
   b) Choose an existing template.  
   The **Phone Button Template Configuration** window appears.

#### 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. See the What to Do Next section for more information.

### What to Do Next

**Associate Direct Transfer Button Template with a Phone**, on page 442
**Associate Direct Transfer Button Template with a Phone**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window is displayed.

**Step 2** From the Find and List Phones window, click Find. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.

**Step 3** Choose the phone to which you want to add the phone button template. The Phone Configuration window appears.

**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. Logical partitioning handling takes place in the following circumstances:</td>
</tr>
<tr>
<td></td>
<td>• When a phone user uses Transfer softkey to transfer the call, the second press of the softkey invokes and processes the Call Transfer feature.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• When the transferred and the transferred destination specifies a PSTN participant.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• Before splitting of the primary and secondary calls, and before joining.</td>
</tr>
</tbody>
</table>

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 Feature

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 Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multilevel Precedence and Preemption (MLPP)</td>
<td>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.</td>
</tr>
</tbody>
</table>

---

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, on page 445.</td>
</tr>
</tbody>
</table>
External Call Transfer Restrictions

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>
</tbody>
</table>
### Configure External Call Transfer Restrictions 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 the Service Parameter for Call Transfer Restrictions, on page 446</td>
<td>Block external calls from being transferred to another external device or number.</td>
</tr>
</tbody>
</table>
| **Step 2** | To configure incoming calls perform the following procedures:  
  - Configure the Clusterwide Service Parameter, on page 448  
  - Configure Gateways for Call Transfer Restrictions, on page 448  
  - Configure Trunks for Call Transfer Restrictions, on page 449 | Configure gateways and trunks as OnNet (internal) or OffNet (external) by using Gateway Configuration or Trunk Configuration or by setting a clusterwide service parameter. |
| **Step 3** | Configure Outgoing Calls, on page 450 | Configure transfer capabilities with route pattern configuration. |

**Configure the Service Parameter for Call Transfer Restrictions**

To block external calls from being transferred to another external device or number:
Procedure

Step 1  From the Cisco Unified CM Administration user interface choose **System > Service Parameters**.

Step 2  On the Service Parameter Configuration window choose the Cisco Unified CM server you want to configure from the Server drop-down menu.

Step 3  Choose **Cisco CallManager (Active)** from the Service drop-down menu.

Step 4  Choose **True** from the Block OffNet to OffNet Transfer drop-down menu. The default value specifies False.

Step 5  Click **Save**.

What to Do Next

Configure the Clusterwide Service Parameter, on page 448

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> (Optional) <strong>Configure the Clusterwide Service Parameter</strong>, on page 448</td>
<td>Configure all gateways or trunks in the Cisco Unified Communications Manager cluster to be OffNet (external) or OnNet (internal).</td>
</tr>
</tbody>
</table>
| **Step 2** **Configure Gateways for Call Transfer Restrictions**, on page 448 | 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 |
| **Step 3** **Configure Trunks for Call Transfer Restrictions**, on page 449 | 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 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 two steps:

Before You Begin

Configure the Service Parameter for Call Transfer Restrictions, on page 446

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>

**Step 1** From the Cisco Unified CM Administration user interface choose **System > Service Parameters**.

**Step 2** On the Service Parameter Configuration window choose the Cisco Unified CM server you want to configure from the Server drop-down menu.

**Step 3** Choose **Cisco CallManager (Active)** from the Service drop-down menu.

**Step 4** Choose either OffNet or OnNet (the default specifies OffNet) from the Call Classification drop-down menu.

What to Do Next

Configure Gateways for Call Transfer Restrictions, on page 448

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 448

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>

**Step 1** From Cisco Unified Communications Manager Administration, choose **Device > Gateway**.
The Find and List Gateways window displays.

**Step 2** To list the configured gateways, click **Find**.
The gateways that are configured in Cisco Unified Communications Manager display.
Step 3 Choose the gateway that you want to configure as OffNet or OnNet.

Step 4 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).

Step 5 Click Save.

What to Do Next

Configure Trunks for Call Transfer Restrictions, on page 449

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 448

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 450
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 449

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.

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 Ad Hoc Conferencing, on page 183.</td>
</tr>
</tbody>
</table>
Interaction Feature

Bulk Administration

Bulk Administration inserts gateway configuration (OffNet or OnNet) on the Gateway Template. For more information, see the *Cisco Unified Communications Manager Bulk Administration Guide*.

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*.

Restrictions

<table>
<thead>
<tr>
<th>Restriction</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FXS Gateways</td>
<td>FXS gateways such as Cisco Catalyst 6000 24 Port do not have a Call Classification field on the Gateway Configuration window; therefore, the system always considers them as OnNet.</td>
</tr>
<tr>
<td>Cisco VG-248 Gateway</td>
<td>The system does not support the Cisco VG-248 Gateway which does not have a Call Classification field.</td>
</tr>
<tr>
<td>FXS Ports</td>
<td>Cisco Unified Communications Manager considers all Cisco Unified IP Phones and FXS ports as OnNet (internal) that cannot be configured as OffNet (external).</td>
</tr>
</tbody>
</table>
PART X

Presence and Privacy Features

- Barge, page 455
- BLF Presence, page 467
- Call Display Restrictions, page 481
- Do Not Disturb, page 495
- Privacy, page 509
- Private Line Automatic Ringdown, page 515
- Secure Tone, page 521
Barge

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 cbarged 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 7.

**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. <strong>Note</strong>: 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>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>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>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>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure Softkey Template for Built-In Conferencing, on page 458</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure Softkey Template for Shared Conferencing, on page 459</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>To Associate a Softkey Template with Common Device Configuration, on page 461, complete the following subtasks:</td>
</tr>
</tbody>
</table>
### Command or Action
- Add a Softkey Template to Common Device Configuration, on page 461
- Associate Common Device Configuration with Phone, on page 462

### Purpose
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 4
**Associate Softkey Template with Phone, on page 460**
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.

### Step 5
**Configure Barge for Built-In Conferencing, on page 462**
Configure barge for built-in conference bridges.

### Step 6
**Configure Barge for Shared Conferencing, on page 463**
Configure barge for shared conference bridges.

### Step 7
**Associate User with Device, on page 49**
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**. The **Softkey Template Configuration** window appears.

#### Step 2
Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.

d) Click **Save**.

#### Step 3
Perform this step to add softkeys to an existing template.

a) Enter search criteria and click **Find**.

b) Choose an existing template.
The Softkey Template Configuration window appears.

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

**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. See the What to Do Next section for more information.

---

**What to Do Next**

Perform one of the following procedures:

- Add a Softkey Template to Common Device Configuration, on page 461
- Associate Common Device Configuration with Phone, on page 462

**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. The Softkey Template Configuration window appears.

**Step 2** Perform this step 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) In the Softkey Template Name field, enter a new name for the template.
- d) Click Save.

**Step 3** Perform this step to add softkeys to an existing template.

- a) Enter search criteria and click Find.
b) Choose an existing template. The **Softkey Template Configuration** window appears.

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

**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. See the What to Do Next section for more information.

---

**What to Do Next**

Perform one of the following procedures:

- **Add a Softkey Template to Common Device Configuration**, on page 461

- **Associate Common Device Configuration with Phone**, on page 462

---

**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 460.

**Procedure**

**Step 1** Add a Softkey Template to Common Device Configuration, on page 342

**Step 2** Associate a Common Device Configuration with a Phone, on page 343

**Add a Softkey Template to Common Device Configuration**

**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Device > Device Settings > Common Device Configuration**. The **Find and List Common Device Configuration** window appears.

**Step 2** Perform this step 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) In the **Name** field, enter a name for the Common Device Configuration.

c) Click **Save**.

**Step 3** Perform this step to add the softkey template to an existing Common Device Configuration.

a) Enter search criteria and click **Find**.

b) Choose an existing Common Device Configuration.

The **Common Device Configuration** window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.

If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.

What to Do Next

Associate Common Device Configuration with Phone, on page 462

Associate Common Device Configuration with Phone

Procedure

Step 1 From Cisco Unified CM Administration, choose Device > Phone.
The Find and List Phones window appears.
Step 2 Find the phone to which 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.

Configure Barge for Built-In Conferencing

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/CBarge 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

**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

### Feature

<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. <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. 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>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</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>
<tr>
<td>G.711 codec</td>
<td>The original call requires G.711 codec. If G.711 is not available, use cBarge instead.</td>
</tr>
<tr>
<td>Cisco Unified IP Phones</td>
<td>You can assign a softkey template that contains the Barge softkey to any IP phone that uses softkeys; however, some IP phones do not support the Barge feature.</td>
</tr>
<tr>
<td>Encryption</td>
<td>If you configure encryption for Cisco Unified IP Phones 7960 and 7940, those encrypted devices cannot accept a barge request when they are participating in an encrypted call. When the call is encrypted, the barge attempt fails. A tone plays on the phone to indicate that the Barge failed.</td>
</tr>
<tr>
<td>Maximum number of calls</td>
<td>If the number of shared-line users in the conference is equal to or greater than the configuration for the Maximum Number of Calls setting for the device from which you are attempting to barge, the phone displays the message, Error: Past Limit.</td>
</tr>
</tbody>
</table>

### Barge Troubleshooting

#### No Conference Bridge Available

**Problem**

When the Barge softkey is pressed, the message *No Conference Bridge Available* is displayed on the IP phone.

**Possible Cause**

The *Built In Bridge* field in the *Phone Configuration* window for the target phone is not set properly.

**Solution**

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

Problem
The phone displays the message, Error: Past Limit.

Possible Cause
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.

Solution
• 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

### Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Configure and synchronize cluster-wide enterprise parameters for BLF. See Configure/Synchronize Cluster-Wide Enterprise Parameters for BLF, on page 469.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Configure cluster-wide service parameters for BLF. See Configure Cluster-Wide Service Parameters for BLF, on page 470.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configure BLF presence groups. See Configure BLF Presence Groups, on page 470.</td>
</tr>
</tbody>
</table>
| **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 472.  
- Associate BLF presence groups with SIP trunks. See Associate BLF Presence Groups with SIP Trunk, on page 473.  
- Associate BLF presence groups with an end user. See Associate BLF Presence Groups with End User, on page 474.  
- Associate BLF presence groups with an application user. See Associate BLF Presence Groups with Application User, on page 475. | 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 475. | To enable application-level authorization for a SIP trunk application in addition to trunk-level authorization. |
### Command or Action | Purpose
--- | ---
**Step 6** | Configure Calling Search Space. See [Configure Calling Search Space](#), on page 476. Apply a SUBSCRIBE Calling Search Space to the SIP trunk, phone, or end user. The SUBSCRIBE Calling Search Space determines how Cisco Unified Communications Manager routes presence requests that come from the trunk or the phone. Calling search spaces determine the partitions that calling devices search when they are attempting to complete a call. If you do not select a different calling search space for presence requests, the SUBSCRIBE Calling Search Space selects the default option, which is **None**.

**Step 7** | Configure a phone button template for BLF and SpeedDial buttons. See [Configure a Phone Button Template for BLF and SpeedDial Buttons](#), on page 477. Configure a phone button template for BLF and SpeedDials for a phone, or user device profile. **Note** If the template does not support BLF and SpeedDials, the Add a new BLF SD link appears in the Unassigned Associated Items pane.

**Step 8** | Associate button template with a device. See [Associate Button Template with a Device](#), on page 478. Use a button template with a configured device for the BLF presence.

**Step 9** | Configure user device profile. See [Configure User Device Profile](#), on page 478. Configure the user device profiles for BLF presence.

---

**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

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

**Step 2** | Configure the fields in the **Enterprise Parameters Configuration** window. See the online help for more information about the fields and their configuration options. **Tip** For details about an enterprise parameter, click the parameter name or the question mark that appears in the **Enterprise Parameter Configuration** window.

**Step 3** | Click **Save**.

**Step 4** | (Optional) Click **Apply Config** to synchronize cluster-wide parameters.

  The Apply Configuration Information dialog box appears.

**Step 5** | Click **OK**.
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 469

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.

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 470

**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 472
- Associate BLF Presence Groups with SIP Trunk, on page 473
- Associate BLF Presence Groups with End User, on page 474
- Associate BLF Presence Groups with Application User, on page 475

**Related Topics**

*BLF Presence Group Fields for BLF*, on page 471

**BLF Presence Group Fields for BLF**

Presence authorization works with BLF presence groups. The following table describes the BLF presence group configuration fields.

<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>
</tbody>
</table>
Modify Relationship to Other Presence Groups | Select one or more BLF presence groups to configure the permission fields for the named group to the selected groups.

Subscription Permission | For the selected BLF presence groups, choose one of the following options from the drop-down list:

- **Use System Default**—Set the permissions field to the Default Inter-Presence Group Subscription clusterwide service parameter field (Allow Subscription or Disallow Subscription).
- **Allow Subscription**—Allow members in the named group to view the real-time status of members in the selected groups.
- **Disallow Subscription**—Block members in the named group from viewing the real-time status of members in the selected groups.

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.

### 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 collocated.

**Before You Begin**

Configure BLF Presence Groups, on page 470

**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 473
- Associate BLF Presence Groups with End User, on page 474
- Associate BLF Presence Groups with Application User, on page 475

**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.

**Tip**  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 470
### 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 472
- Associate BLF Presence Groups with End User, on page 474
- Associate BLF Presence Groups with Application User, on page 475

---

### 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 470

#### 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 472
- Associate BLF Presence Groups with SIP Trunk, on page 473
- Associate BLF Presence Groups with Application User, on page 475

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 470

Procedure

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>In the Cisco Unified CM Administration, choose User Management &gt; Application User, and click Add New. The Application User Configuration window appears.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Configure the fields in the Application User Configuration window. See the online help for 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

Associate BLF presence group with devices and users by performing the following subtasks:

- Associate BLF Presence Groups with Phone, on page 472
- Associate BLF Presence Groups with SIP Trunk, on page 473
- Associate BLF Presence Groups with End User, on page 474

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 472
- Associate BLF Presence Groups with SIP Trunk, on page 473
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 Calling Search Space, on page 476

Configure Calling Search Space

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.

To apply a SUBSCRIBE Calling Search Space to the SIP trunk, phone, or end user, perform the following procedure:

Before You Begin
Accept BLF Presence Requests from External Trunks and Applications, on page 475
**Procedure**

**Step 1**  
To configure calling search space, in Cisco Unified CM Administration, click **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 about 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 477

---

**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 Calling Search Space, on page 476
### 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. For example, BLF SIP 7970.

**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 478

---

### 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 477

---

### 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 478

---

### Configure User Device Profile

See the “BLF Presence with Extension Mobility” section of BLF Presence Interactions, on page 479 for details.
**Before You Begin**

Associate Button Template with a Device, on page 478

**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 and Restrictions**

**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>
<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 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.</td>
</tr>
<tr>
<td>BLF Presence with Extension Mobility</td>
<td>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.</td>
</tr>
</tbody>
</table>
### 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 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>Because 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. 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>
<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, 7940, 7905, and 7912 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>
Call Display 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

<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 7</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, on page 482.</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 configure a partition for dialing between rooms, and a partition for dialing the public switched telephone network (PSTN).</td>
</tr>
</tbody>
</table>
### Configure Partitions

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.

#### 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 Communications Manager Administration, choose Call Routing &gt; Class of Control &gt; Partition.</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>Click Add New to create a new partition.</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>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 Related Topics section for guidelines about partition names.</td>
<td></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>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>To create multiple partitions, use one line for each partition entry.</td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td>From the Time Schedule drop-down list, choose a time schedule to associate with this partition.</td>
<td></td>
</tr>
</tbody>
</table>
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 Calling Search Space for External Call Control, on page 483

**Related Topics**

Partition Name Guidelines, on page 483

---

**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>
<thead>
<tr>
<th>Partition Name Length</th>
<th>Maximum Number of Partitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>2 characters</td>
<td>170</td>
</tr>
<tr>
<td>3 characters</td>
<td>128</td>
</tr>
<tr>
<td>4 characters</td>
<td>102</td>
</tr>
<tr>
<td>5 characters</td>
<td>86</td>
</tr>
<tr>
<td>...</td>
<td>...</td>
</tr>
<tr>
<td>10 characters</td>
<td>46</td>
</tr>
<tr>
<td>15 characters</td>
<td>32</td>
</tr>
</tbody>
</table>

**Configure a Calling Search Space for External Call Control**

Configure a calling search space that Cisco Unified Communications Manager uses when the route server sends a divert obligation to Cisco Unified Communications Manager. 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.

**Before You Begin**

Set up the Cisco Unified Routing Rules Interface so that the route server can direct Cisco Unified Communications Manager on how to handle calls.


**Procedure**

**Step 1** From Cisco Unified CM Administration, select **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 an External Call Control Profile

**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
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.

What to Do Next
Configure Translation Patterns, on page 485

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 484

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 487

Related Topics
Translation Pattern Fields for Call Display Restrictions, on page 486
### Translation Pattern Fields for Call Display Restrictions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Translation Pattern</strong></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><strong>Description</strong></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><strong>Partition</strong></td>
<td>From the drop-down list, choose the partition to associate with this translation pattern.</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong></td>
<td>From the drop-down list, choose the calling search space to associate with this translation pattern.</td>
</tr>
<tr>
<td><strong>Calling Line ID Presentation</strong></td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td>• <strong>Default</strong>—Choose this option if you do not want to change the presentation of the calling line ID.</td>
<td></td>
</tr>
<tr>
<td>• <strong>Allowed</strong>—Choose this option if you want to display the phone number of the calling party.</td>
<td></td>
</tr>
<tr>
<td>• <strong>Restricted</strong>—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling party phone number.</td>
<td></td>
</tr>
<tr>
<td><strong>Calling Name Presentation</strong></td>
<td>From the drop-down list, choose one of the following options:</td>
</tr>
<tr>
<td>• <strong>Default</strong>—Choose this option if you do not want to change the presentation of the calling name.</td>
<td></td>
</tr>
<tr>
<td>• <strong>Allowed</strong>—Choose this option if you want to display the name of the calling party.</td>
<td></td>
</tr>
<tr>
<td>• <strong>Restricted</strong>—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling name.</td>
<td></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 485

**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**.

<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. |
| **Connected 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 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. |
The **Add a New Phone** 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 a dialed number.

**Step 4**  Check the **Ignore presentation indicators (internal calls only)** check box to ignore any presentation restriction on internal calls.

**Step 5**  Click **Save**.  
The phone is added to the database.

**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_Phones, CSS_FromRoom} : 221/Room-1

Configure phone B (Room-2) with partition P_Room and device/line calling search space CSS_FromRoom

{ P_Phones, 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 488

---

**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 487

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 489

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 488

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.
## SIP Trunk Fields for Call Display Restrictions

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Inbound Calls</strong></td>
<td></td>
</tr>
</tbody>
</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.                                           |
| **Outbound Calls**    |                                                                                                                                               |
| **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. |
**Field** | **Description**  
---|---  
Connected 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 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.
Interaction Feature

**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.

   Call Park and Directed Call Park, on page 347

---

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Conference List</strong></td>
<td>When you use Call Display Restrictions, you restrict the display information for the list of participants in a conference.</td>
</tr>
<tr>
<td></td>
<td><em>Ad Hoc Conferencing</em>, on page 183</td>
</tr>
<tr>
<td><strong>Conference and Voice Mail</strong></td>
<td>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 <strong>To Conference</strong>. When voice mail is accessed by choosing the Messages button, the call information display shows <strong>To Voicemail</strong>.</td>
</tr>
<tr>
<td>Feature</td>
<td>Interaction</td>
</tr>
<tr>
<td>-----------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Extension Mobility</td>
<td>To use Call Display Restrictions with Extension Mobility, enable the <strong>Ignore Presentation Indicators (internal calls only)</strong> 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).</td>
</tr>
<tr>
<td>Call Forwarding</td>
<td>The Connected Number Display restriction applies to all calls that originate in the system. When this value is set to <strong>True</strong>, 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.</td>
</tr>
</tbody>
</table>

**Call Display Restrictions Feature Restrictions**

Translation Patterns—Duplicate entries are not allowed in translation patterns.
Do Not Disturb

This chapter provides details on the Do Not Disturb feature.

- Do Not Disturb Overview, page 495
- Do Not Disturb Configuration Task Flow, page 496
- Interactions and Restrictions, page 504
- Troubleshooting Do Not Disturb, page 507

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

Note

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>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Run a Phone Feature List report from Cisco Unified Reporting to determine which phones support Do Not Disturb. Note Cisco Unified IP Phones 7940 and 7960 that are running SIP use their own backwards-compatible implementation of Do Not Disturb, which you configure on the SIP Profile.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

| Step 2 | Configure Busy Lamp Field Status, on page 497 | Configure the Busy Lamp Field status service parameter. |

| Step 3 | Configure Do Not Disturb on a Common Phone Profile, on page 497 | 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. |

| Step 4 | Apply Do Not Disturb Settings to the Phone, on page 498. | Apply Do Not Disturb settings to the phone. |

| Step 5 | Depending on whether your phone uses softkeys or feature buttons, perform either of the following tasks: • Configure a Do Not Disturb Feature Button, on page 499 | Add a Do Not Disturb feature button or softkey to your phone. |
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:

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose System &gt; Service Parameters.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Choose the Cisco CallManager service for the server that you want to configure.</td>
</tr>
<tr>
<td>Step 3</td>
<td>In the Clusterwide Parameters (System - Presence) pane, specify one of the following values for the BLF Status Depicts DND service parameter:</td>
</tr>
<tr>
<td></td>
<td>• <em>True</em>—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.</td>
</tr>
<tr>
<td></td>
<td>• <em>False</em>—If Do Not Disturb is activated on the device, the BLF status indicator for the device or line appearance reflects the actual device state.</td>
</tr>
</tbody>
</table>

**What to Do Next**

Perform one of the following procedures:

- Configure Do Not Disturb on a Common Phone Profile, on page 497
- Apply Do Not Disturb Settings to the Phone, on page 498

**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 497
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 **Incoming 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 498

---

**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 497.

Otherwise, complete Configure Busy Lamp Field Status, on page 497
**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.

**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 499
- **Configure a Do Not Disturb Softkey**, on page 501

**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>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Configure Phone Button Template for Do Not Disturb, on page 500</td>
<td>Create a phone button template that includes the Do Not Disturb button.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Associate a Button Template with a Phone, on page 211</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. The Find and List Phone Button Templates window appears.

**Step 2**
Click Find. The window displays a list of templates for the supported phones.

**Step 3**
Perform this step 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 this step if you want to add phone buttons to an existing template.
   a) Enter search criteria and click Find.
   b) Choose an existing template.
   The Phone Button Template Configuration window appears.

**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. See the What to Do Next section for more information.

---

**What to Do Next**

*Associate a Button Template with a Phone, on page 211*
Associate a Button Template with a Phone

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>From Cisco Unified CM Administration, choose Device &gt; Phone. The Find and List Phones window is displayed.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>From the Find and List Phones window, click Find. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Choose the phone to which you want to add the phone button template. The Phone Configuration window appears.</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 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.

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 501</td>
</tr>
</tbody>
</table>
| **Step 2** | Perform either of the following procedures:  
  • Associate a Softkey Template with a Common Device Configuration, on page 502  
  • Associate Softkey Template with a Phone, on page 504 |

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**. The **Softkey Template Configuration** window appears.

**Step 2** Perform this step 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) In the **Softkey Template Name** field, enter a new name for the template.

d) Click **Save**.

**Step 3** Perform this step to add softkeys to an existing template.

a) Enter search criteria and click **Find**.

b) Choose an existing template.

The **Softkey Template Configuration** window appears.

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

**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. See the What to Do Next section for more information.

**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 502

*Associate Softkey Template with a Phone*, on page 504

**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 501

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Add Softkey Template to Common Device Configuration, on page 503</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Associate Common Device Configuration with Phone, on page 504</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. The Find and List Common Device Configuration window appears.

**Step 2** Perform this step 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) In the Name field, enter a name for the Common Device Configuration.
   c) Click Save.

**Step 3** Perform this step to add the softkey template to an existing Common Device Configuration.
   a) Enter search criteria and click Find.
   b) Choose an existing Common Device Configuration. The Common Device Configuration window appears.

**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 created a new Common Device Configuration, associate the configuration with devices and then restart them. See the What to Do Next section for more information.
   • If you modified a Common Device Configuration that is already associated with devices, click Apply Config to restart the devices.

What to Do Next

Associate Common Device Configuration with Phone, on page 504
Associate Common Device Configuration with Phone

**Before You Begin**

Associate a Softkey Template with a Common Device Configuration, on page 502

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**
Find the phone to which 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 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 501

**Procedure**

**Step 1**
From Cisco Unified CM Administration, choose **Device > Phone**. The **Find and List Phones** window appears.

**Step 2**
Choose the phone to which you want to add the softkey template. The **Phone Configuration** window appears.

**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.

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.

---

**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. |
## Interactions and Restrictions

### Feature Configuration Guide for Cisco Unified Communications Manager, Release 10.5(2)

**Feature** | **Interaction with Do Not Disturb**
--- | ---
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
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

Troubleshooting Do Not Disturb

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>
<tbody>
<tr>
<td>DND softkey does not display or</td>
<td>- Verify that the softkey or button template for this phone includes</td>
</tr>
<tr>
<td>DND feature button does not display</td>
<td>DND.</td>
</tr>
<tr>
<td></td>
<td>- Capture a sniffer trace and verify that the phone gets the correct</td>
</tr>
<tr>
<td></td>
<td>softkey or button template.</td>
</tr>
<tr>
<td></td>
<td>- Verify that the phone firmware is Version 8.3(1) or later.</td>
</tr>
<tr>
<td>BLF speed dial does not show DND status</td>
<td>- Verify that the BLF DND is set to enabled in Enterprise parameters.</td>
</tr>
<tr>
<td></td>
<td>- Capture a sniffer trace and verify that the phone gets the correct</td>
</tr>
<tr>
<td></td>
<td>notification message.</td>
</tr>
<tr>
<td></td>
<td>- Verify that the phone firmware is Version 8.3(1) or later.</td>
</tr>
</tbody>
</table>
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 7</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 510</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 510</td>
<td>Enable Privacy for specific devices.</td>
</tr>
<tr>
<td>Step 4 Configure Privacy Phone Button Template, on page 511</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 512</td>
<td>Associate the phone button template with a user.</td>
</tr>
<tr>
<td>Step 6 Configure Shared Line Appearance, on page 512</td>
<td>Configure the shared line appearance.</td>
</tr>
<tr>
<td>Step 7 (Optional) Configure Privacy on Hold, on page 513</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 7**.
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 511**

Configure Privacy Phone Button Template

Procedure

Step 1  From Cisco Unified CM Administration, choose **Device > Device Settings > Phone Button Template**. The **Find and List Phone Button Templates** window appears.
Step 2  Click **Find**. The window displays a list of templates for the supported phones.
Step 3  Perform this step 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 this step if you want to add phone buttons to an existing template.
   a)  Enter search criteria and click **Find**.
   b)  Choose an existing template. The **Phone Button Template Configuration** window appears.
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. See the What to Do Next section for more information.

What to Do Next

**Associate Privacy Phone Button Template with a Phone, on page 512**
**Associate Privacy Phone Button Template with a Phone**

**Before You Begin**
Configure Privacy Phone Button Template, on page 511

**Procedure**

1. From Cisco Unified CM Administration, choose Device > Phone. The Find and List Phones window is displayed.
2. From the Find and List Phones window, click Find. A list of phones that are configured on the Cisco Unified Communications Manager is displayed.
3. Choose the phone to which you want to add the phone button template. The Phone Configuration window appears.
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 Shared Line Appearance**

**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 512 to Step 6, on page 512 for all the phones for which you want to create a shared line appearance.

**Note**
Ensure that you assign the same directory number and route partition to all the phones that are part of the 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 System &gt; Service Parameters. The Service Parameter Configuration window appears.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>From the Server 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 Service drop-down list, choose Cisco CallManager.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Set the Enforce Privacy Setting on Held Calls service parameter to True.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Click Save.</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

This chapter describes how to configure Private Line Automatic Ringdown.

- Private Line Automatic Ringdown Overview, page 515
- Private Line Automatic Ringdown Configuration Task Flow for SCCP Phones, page 515
- Private Line Automatic Ringdown Configuration Task Flow for SIP Phones, page 518
- Private Line Automatic Ringdown Troubleshooting, page 519

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</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Create Partition, on page 516</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 516</td>
<td>Assign the partition to a unique CSS, and a CSS that includes the PLAR destination device.</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**.

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

Complete the **Create Partition, on page 516** task.
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.

Assign Partition to the Private Line Automatic Ringdown Destination

When configuring Private Line Automatic Ringdown (PLAR) on SCCP phones, you must 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.

Procedure

Step 1 In Cisco Unified CM Administration, choose Call Routing > Directory Number.
Step 2 Click Find and select the directory number that you want to use as the PLAR destination.
Step 3 In the Route Partition field, select a partition that you created for your PLAR destination.
Step 4 In the Calling Search Space drop-down list box, select the CSS that includes both the null partition and the destination device.
Step 5 Click Save.

What to Do Next
Configure Translation Pattern for Private Line Automatic Ringdown on SCCP Phones, on page 517

Configure Translation Pattern for Private Line Automatic Ringdown on SCCP Phones

To configure Private Line Automatic Ringdown (PLAR) on SCCP phones, you must 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 517
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>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Create SIP Dial Rule for Private Line Automatic Ringdown, on page 518</td>
<td>Create a SIP dial rule for PLAR.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Assign Private Line Automatic Ringdown Dial Rule to SIP Phone, on page 519</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.
Procedure

Step 1  In Cisco Unified CM Administration, choose Call Routing > Class of Control > SIP Dial Rules.
Step 2  Click Add New.
Step 3  From the Dial Pattern drop-down list box, choose 7940_7960_OTHER.
Step 4  Click Next.
Step 5  Enter a name and description for the dial rule.
Step 6  Click Next.
Step 7  In the Pattern field, enter a pattern that matches the PLAR destination number and click Add PLAR.
Step 8  Click Save.

What to Do Next

Assign Private Line Automatic Ringdown Dial Rule to SIP Phone, on page 519

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 518

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>Symptom</td>
<td>Solution</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>---------------------------------------------------------------------------</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 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**

<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 7</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 523</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 523</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 524</td>
<td>Configure service parameters.</td>
</tr>
<tr>
<td>Step 5</td>
<td>(Optional) Configure MGCP E1 PRI Gateway, on page 524</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 7

**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. The Find and List Phones window is displayed.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Click the phone for which you want to set secure tone parameters. The Phone Configuration window is displayed.</td>
</tr>
<tr>
<td>Step 3</td>
<td>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 softkey template without supplementary service softkeys for a protected phone.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Set the Join Across Lines option to Off.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Check the Protected Device check box.</td>
</tr>
<tr>
<td>Step 6</td>
<td>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 &gt; Security Profile &gt; Phone Security Profile).</td>
</tr>
<tr>
<td>Step 7</td>
<td>Click Save.</td>
</tr>
</tbody>
</table>

**What to Do Next**

Perform one of the following procedures:

- Configure Directory Number for Secure Tones, on page 523
- Configure MGCP E1 PRI Gateway, on page 524

Configure Directory Number for Secure Tones

**Before You Begin**

Configure Phone As a Protected Device, on page 523

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Locate the Association section on the Phone Configuration window.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Select Add a new DN.</td>
</tr>
</tbody>
</table>
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.

**Step 5** 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 6** Click Save.

---

**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 523

**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

• Client Matter Codes and Forced Authorization Codes, page 529
• Custom Phone Rings, page 537
• Music On Hold, page 541
• Self Care Portal, page 573
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 to specify 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

- Mobile phones with Cisco Jabber installed that support CMCs and FACs.
- 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, so the feature prompts the user to enter the FAC after the first tone and enter the CMC after the second tone.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 1 | To Configure Client Matter Codes, on page 530, complete the following subtasks:  
• Add Client Matter Codes, on page 530  
• Enable Client Matter Codes, on page 531 | 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. |

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 2 | To Configure Forced Authorization Codes, on page 531, complete the following subtasks:  
• Add Forced Authorization Codes, on page 532  
• Enable Forced Authorization Codes, on page 532 | 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. |

**Configure Client Matter Codes**

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Add Client Matter Codes, on page 530</td>
<td>Determine unique client matter codes that you want to use and add them to Unified Communications Manager. Because the number of CMCs directly affects the time that is required for Cisco Unified Communications Manager 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>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Enable Client Matter Codes, on page 531</td>
<td>Enable client matter codes to enable the feature through a route pattern.</td>
</tr>
</tbody>
</table>

**Add Client Matter Codes**

Because the number of CMCs directly affects the time that is required for Cisco Unified Communications Manager 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.
**Procedure**

**Step 1** From Cisco Unified CM Administration, choose **Call Routing** > **Client Matter Codes**.

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

**Step 3** In the **Client Matter Code** field, enter a unique code of no more than 16 digits that the user will enter when placing a call.

**Step 4** In the **Description** field, enter a client name if you want to associate a client name with the client matter code.

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

**What to Do Next**

Enable Client Matter Codes, on page 531

---

**Enable Client Matter Codes**

**Before You Begin**

Add Client Matter Codes, on page 530

**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><strong>Step 1</strong> Add Forced Authorization Codes, on page 532</td>
<td>Determine unique forced authorization codes that you want to use and add them to Unified Communications Manager. 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.</td>
</tr>
</tbody>
</table>
### Add Forced Authorization Codes

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**

1. From Cisco Unified CM Administration, choose Call Routing > Forced Authorization Codes.
2. In the Authorization Code Name field, enter a unique name that is no more than 50 characters. This name ties the authorization code to a specific user or group of users.
3. In the Authorization Code field, enter a unique authorization code that is no more than 16 digits. The user enters this code when the user places a call through an FAC-enabled route pattern.
4. In the Authorization Level field, enter a three-digit authorization level in the range of 0 to 255.
5. Click Save.

### What to Do Next

Enable Forced Authorization Codes, on page 532

---

### Enable Forced Authorization Codes

**Before You Begin**

Enable Forced Authorization Codes, on page 532

**Procedure**

1. In Cisco Unified CM Administration, choose Call Routing > Route/Hunt > Route Pattern.
2. Perform one of the following tasks:
   - To update an existing route pattern, enter search criteria, click Find, and then 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 Forced Authorization Code** check box.

**Step 4**  In the **Authorization Level** 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.

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

---

### Client Matter Codes and Forced Authorization Codes Interactions and Restrictions

**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 CMCs, FACs, and authorization levels.</td>
</tr>
<tr>
<td>CTI, JTAPI, and TAPI applications</td>
<td>In most cases, Cisco Unified Communications Manager 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 Cisco Unified Communications Manager receives the appropriate codes, the call connects to the intended party. If Cisco Unified Communications Manager does not receive the appropriate codes, Cisco Unified Communications Manager sends an error to the application that indicates which code is missing.</td>
</tr>
</tbody>
</table>
Interaction

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.

Redundancy

Cisco Unified Communications Manager provides redundancy, which handles the normal processes that are in place for Cisco Unified Communications Manager.

Speed Dial and Abbreviated Speed Dial

You can use Speed Dial to reach destinations that require a Forced Authorization Code (FAC), Client Matter Code (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.

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><strong>Restriction</strong></td>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>-----------------------------------</td>
<td>-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</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 <strong>Mobility Identity</strong> window.</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 <strong>Require Forced Authorization Code</strong> or the <strong>Require Client Matter Code</strong> check box in the <strong>Route Pattern Configuration</strong> window, the <strong>Allow Overlap Sending</strong> 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 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 two 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 call back 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 539.

• The Ringlist.xml file must meet a set of formatting guidelines. For details, review the topic **Ringlist.xml File Format Requirements**, on page 539.

**Custom Phone Rings 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>Prepare Custom Phone Rings for Upload, on page 538</td>
<td>Create your customized PCM and Ringlist.xml files.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Upload Custom Phone Rings to TFTP Server, on page 538</td>
<td>Upload customized files to the Cisco Unified Communications Manager TFTP server.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Restart TFTP Service, on page 539</td>
<td>After the upload completes, restart the Cisco TFTP service.</td>
</tr>
</tbody>
</table>

**Prepare Custom Phone Rings for Upload**

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Use the <code>file get tftp &lt;tftp path&gt;</code> CLI command to download the existing Ringlist.xml file, in addition to any PCM files that you want to modify.</td>
</tr>
<tr>
<td>Step 2</td>
<td>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 <strong>PCM File Format Requirements</strong>, on page 539.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Use an ASCII editor to update the Ringlist.xml file with your new phone rings. For details on Ringlist.xml file formatting requirements, see <strong>Ringlist.xml File Format Requirements</strong>, on page 539.</td>
</tr>
</tbody>
</table>

**What to Do Next**

Upload Custom Phone Rings to TFTP Server, on page 538

**Upload Custom Phone Rings to TFTP Server**

**Before You Begin**

Prepare Custom Phone Rings for Upload, on page 538
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 539](#)

**Restart TFTP Service**

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/>
    <FileName/>
  </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.
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).

Music On Hold Prerequisites

- A Cisco Unified Communications Manager system that is configured to use the Music On Hold (MOH) streams that the MOH server provides when a call is placed on hold.
- 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.

Music On Hold Configuration Task Flow

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Enable Music On Hold. See Enable Music on Hold, on page 544.</td>
<td>Enable the Music On Hold service.</td>
</tr>
</tbody>
</table>
| Step 3 | Configure MOH audio. See Music On Hold Audio Source Configuration, on page 548, and perform the following subtasks:  
  - Convert MOH Files. See Convert Music On Hold Files, on page 549.  
  - Configure MOH audio source. See Configure Music on Hold Audio Source, on page 549. | • Upload a Music On Hold audio file to make it available for use as a Music On Hold audio source.  
  • Convert the Music On Hold file to the appropriate formats for use by the Music On Hold server.  
  • To place on-net and off-net users on hold (end user hold or network hold) with music streamed from a streaming source. |
<p>| Step 4 | (Optional) Configure fixed MOH audio source. See Configure Fixed Music On Hold Audio Source, on page 553. | Configure the fixed MOH audio source in addition to the file stream sources. |
| Step 6 | Configure Media Resource Group list. See Configure Media Resource Group List, on page 556. | Specify a list of prioritized media resource groups. |
| Step 7 | View MOH audio file. See View Music on Hold Audio File, on page 557. | View a list of Music On Hold audio files that are stored on the system. |</p>
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 8</strong> Enable security for MOH. See Enable Security for Music On Hold, on page 558.</td>
<td>Enable security for Music On Hold devices through the <strong>Cluster Security Mode</strong> enterprise parameter.</td>
</tr>
<tr>
<td><strong>Step 9</strong> (Optional) Enable secured MOH through SRTP. See Enable Secured Music On Hold through SRTP, on page 559.</td>
<td>When you enable the Cisco Unified Communications Manager cluster or system for security, the MOH server registers with the Cisco Unified Communications Manager as an SRTP-capable device.</td>
</tr>
<tr>
<td><strong>Step 10</strong> Configure multicast by performing the following subtasks:</td>
<td>Configure the various Cisco Unified Communications Manager services to allow multicasting. For details on unicast and multicast audio sources, see <strong>Unicast and Multicast Audio Sources</strong>, on page 560.</td>
</tr>
<tr>
<td>• Plan MOH Server capacity. See Plan Music On Hold Server Capacity, on page 561.</td>
<td></td>
</tr>
<tr>
<td>• Verify Music On Hold service parameters. See Verify Music On Hold Service Parameters, on page 562.</td>
<td></td>
</tr>
<tr>
<td>• Configure multicast Music On Hold audio sources or fixed MOH audio source. See Configure Multicast Music On Hold Audio Sources/Fixed MOH Audio Source, on page 563.</td>
<td></td>
</tr>
<tr>
<td>• Configure a multicast-enabled media resource group. See Configure a Multicast-Enabled Media Resource Group, on page 564.</td>
<td></td>
</tr>
<tr>
<td>• Configure multicast Music On Hold over H.323 intercluster trunks. See Configure Multicast Music On Hold over H.323 Intercluster Trunks, on page 565.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 11</strong> (Optional) Reset or restart a Music On Hold server. See Reset or Restart a Music On Hold Server, on page 566.</td>
<td>Reset or restart a music on hold server for changes to take effect, if required.</td>
</tr>
<tr>
<td><strong>Step 12</strong> (Optional) Synchronize Music On Hold server. See Synchronize Music On Hold Server, on page 566.</td>
<td>Apply configuration to the selected music on hold servers.</td>
</tr>
</tbody>
</table>
Enable Music on Hold

When you install Cisco Unified Communications Manager, the Cisco IP Voice Media Streaming application is installed automatically. However, you need to enable the Music On Hold feature to use it.

**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** In Cisco Unified Serviceability, choose Application > Serviceability Webpage, and enter a valid username and password.

**Step 2** Choose Tools > Service Activation. The Service Activation window appears.

**Step 3** Choose a server from the Server drop-down list.

**Step 4** From the CMServices section, check the Cisco IP Voice Media Streaming App check box. The Music On Hold service is enabled.

**What to Do Next**

Configure Music On Hold Server, on page 544

Configure Music On Hold Server

**Before You Begin**

- Enable Music on Hold, on page 544.
- 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** In the Cisco Unified CM Administration, choose Media Resources > Music On Hold Server. The Find and List Music On Hold Servers window appears.

**Step 2** Choose the two drop-down list boxes to search for a music on hold server.

**Step 3** Choose the Music On Hold server that you want to update.
The **Music On Hold (MOH) Server Configuration** window appears.

**Step 4**
Configure the fields from the **Music On Hold (MOH) Server Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

**Step 5**
Click **Save**.
The Music On Hold server is updated in the database. When a server is updated, Cisco Unified Communications Manager adds the media termination point, conference bridge, annunciator, and Music On Hold devices to the database.

---

**What to Do Next**
Configure Music On Hold Audio. Perform the following procedures:

- Upload Music On Hold Audio File, on page 548
- Convert Music On Hold Files, on page 549
- Configure Music on Hold Audio Source, on page 549

**Related Topics**
Music On Hold Server Fields for Music On Hold, on page 545

---

**Music On Hold Server Fields for Music On Hold**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Device Information</strong></td>
<td></td>
</tr>
<tr>
<td>Registration</td>
<td>Displays the registration information of the device.</td>
</tr>
<tr>
<td>IPv4 Address</td>
<td>Displays the IPv4 address.</td>
</tr>
<tr>
<td>IPv6 Address</td>
<td>Displays the IPv6 address.</td>
</tr>
<tr>
<td>Device is trusted</td>
<td>If the device is trusted, a green checkmark appears.</td>
</tr>
<tr>
<td>Host Server</td>
<td>Displays the IP address of the existing host server.</td>
</tr>
<tr>
<td>Music On Hold Server Name</td>
<td>Enter a unique name for the MOH server. The name can comprise up to 15 characters. You can form the name by using letters, numbers, spaces, dashes, dots (periods), and underscores.</td>
</tr>
<tr>
<td>Description</td>
<td>Enter a description for the MOH server. The description can include up to 50 characters. Ensure that this field does not contain ampersand (&amp;), double quotes (&quot;), brackets ([]), less than (&lt;), greater than (&gt;), or the percentage (%).</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Choose a device pool for the Music On Hold server from the drop-down arrow and choose a device pool from the list that appears.</td>
</tr>
</tbody>
</table>

---
Use locations to implement call admission control (CAC) in a centralized call-processing system. CAC enables you to regulate audio quality and video availability by limiting the amount of bandwidth that is available for audio and video calls over links between locations. The location specifies the total bandwidth that is available for calls to and from this location.

From the drop-down list, choose the appropriate location for this MOH server.

The **Hub_None** location field indicates that the locations feature does not keep track of the bandwidth that this MOH server consumes. The **Phantom** location field indicates a location that enables successful CAC across intercluster trunks that use H.323 or SIP protocol.

To configure a new location, use the **System > Location** menu option.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Location</td>
<td>Use locations to implement call admission control (CAC) in a centralized call-processing system. CAC enables you to regulate audio quality and video availability by limiting the amount of bandwidth that is available for audio and video calls over links between locations. The location specifies the total bandwidth that is available for calls to and from this location. From the drop-down list, choose the appropriate location for this MOH server. The <strong>Hub_None</strong> location field indicates that the locations feature does not keep track of the bandwidth that this MOH server consumes. The <strong>Phantom</strong> location field indicates a location that enables successful CAC across intercluster trunks that use H.323 or SIP protocol. To configure a new location, use the <strong>System &gt; Location</strong> menu option.</td>
</tr>
<tr>
<td>Maximum Half Duplex Streams</td>
<td>Enter a number in this required field for the maximum number of unicast Music On Hold streams that this Music On Hold server supports. This value determines 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. Valid values range from 0 to 1000.</td>
</tr>
<tr>
<td>Maximum Multi-cast Connections</td>
<td>Enter a number in this required field for the maximum number of multicast Music On Hold streams that this Music On Hold server supports. This value determines the maximum number of devices that can be on multicast music on hold that is streamed from this Music On Hold server at any given time. Valid values range from 1 to 999999.</td>
</tr>
<tr>
<td>Fixed Audio Source Device</td>
<td>Enter the device name of the fixed audio source device. This device serves as the per-server override that is used if the server has a special sound device installed.</td>
</tr>
</tbody>
</table>
**Field** | **Description**
--- | ---
Use Trusted Relay Point | From the drop-down list, enable or disable whether Cisco Unified Communications Manager inserts a trusted relay point (TRP) device with this media endpoint. Choose one of the following values:
  - **Off**—Disables the use of a TRP with this device.
  - **On**—Enables the use of a TRP with this device.
  
  A trusted relay point (TRP) device designates an Media Transfer Protocol (MTP) or transcoder device that is labeled as Trusted Relay Point.

  Cisco Unified Communications Manager places the TRP closest to the associated endpoint device if more than one resource is needed for the endpoint (for example, a transcoder or RSVP Agent).

  If both TRP and MTP are required for the endpoint, TRP is used as the required MTP.

  If both TRP and RSVP Agent are needed for the endpoint, Cisco Unified Communications Manager first tries to find an RSVP Agent that can also be used as a TRP.

  If both TRP and transcorder are needed for the endpoint, Cisco Unified Communications Manager first tries to find a transcoder that is also designated as a TRP.

Run Flag | Use this required field to choose a run flag for the Music On Hold server. To do so, click the drop-down arrow and choose Yes or No. Choosing No disables the music on hold server.

**Multicast Audio Source Information**

Enable Multicast Audio Sources on this MOH Server | Check or uncheck this check box to enable or disable the multicast of audio sources for this Music On Hold server.

**Note** If this MOH server belongs to a multicast media resource group, a message asks you to enable multicast on this MOH server or to update the specified media resource groups either by removing this MOH server or by changing the multicast field of each listed group.

Base Multicast IP Address | If multicast support is needed, enter the base multicast IP address in this field. Valid IP addresses for multicast range from 224.0.1.0 to 239.255.255.255.

**Note** IP addresses between 224.0.1.0 and 238.255.255.255 are in the reserved range of IP multicast addresses for public multicast applications. Use of these addresses may interfere with existing multicast applications on the Internet. We strongly recommend using IP addresses that are in the range that is reserved for administratively controlled applications on private networks (239.0.0.0 – 239.255.255.255).
If multicast support is needed, enter the base multicast port number in this field. Valid multicast port numbers include even numbers that range from 16384 to 32766.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Base Multicast Port Number</td>
<td>If multicast support is needed, enter the base multicast port number in this field. Valid multicast port numbers include even numbers that range from 16384 to 32766.</td>
</tr>
<tr>
<td>Increment Multicast on</td>
<td>Click <strong>Port Number</strong> to increment multicast on port number. Click <strong>IP Address</strong> to increment multicast on IP address.</td>
</tr>
<tr>
<td>Selected Multicast Audio Sources</td>
<td>This field designates Music On Hold audio stream number that is associated with a particular multicast audio source. Only audio sources that are defined as allowing multicasting appear.</td>
</tr>
<tr>
<td>No.</td>
<td>This field designates the name of the audio source that is defined to allow multicasting.</td>
</tr>
<tr>
<td>Audio Source Name</td>
<td>This field designates the name of the audio source that is defined to allow multicasting.</td>
</tr>
<tr>
<td>Max Hops</td>
<td>For each multicast audio source, enter the maximum number of router hops through which multicast packets should pass. Valid values range from 1 to 127.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Use multicast by incrementing IP address as the preferred method in firewall situations. This results in a unique IP address for each multicast audio source and helps to avoid network saturation.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Using high values can lead to network saturation. This field also gets identified as Time to Live.</td>
</tr>
</tbody>
</table>

### Music On Hold Audio Source Configuration

**Upload Music On Hold Audio File**

When you upload an audio file, it is available for use as a Music On Hold audio source. If you use the Media Resources > Music On Hold Audio Source menu option to add a new audio source, the addition makes the newly uploaded audio file available in the MOH Audio Source File drop-down list.

**Note**

You must upload Music On Hold audio source files to each MOH server.

**Before You Begin**

Configure Music On Hold Server, on page 544

**Procedure**

**Step 1**

In the Cisco Unified CM Administration, choose Media Resources > MOH Audio File Management.
The Music On Hold Audio File Management window appears.

**Step 2** Click **Upload File**.
The Upload File popup window appears.

**Step 3** If you know the path to a file that specifies an audio file, enter the path in the **File** field. If you do not know the path and file name, search for the audio file by clicking **Browse** to the right of the File field. After you find the audio file, click the desired audio file and click **Open**.
The path to the chosen audio file appears in the **File** field of the **Upload File** popup window.

**Step 4** Click **Upload** to upload the specified audio file.
After the audio file gets uploaded, the Upload Result window shows the result of the upload.

**Note** The uploading procedure uploads the file to the Cisco Unified Communications Manager server and performs audio conversions to create codec-specific audio files for MOH. Depending on the size of the original file, processing may take several minutes to complete.

**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 or each 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.

**Step 5** (Optional) Click **Close** to close the **Upload Result** window.

---

**What to Do Next**

- Convert Music On Hold Files, on page 549
- Configure Music on Hold Audio Source, on page 549

---

**Convert Music On Hold Files**

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.

These are some examples of a 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

**Before You Begin**

Upload Music On Hold Audio File, on page 548

**What to Do Next**

Configure Music on Hold Audio Source, on page 549

**Configure Music on Hold Audio Source**

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.
If a new version of an audio source file is available, perform the update procedure to use the new version.

**Before You Begin**

Convert Music On Hold Files, on page 549

**Procedure**

**Step 1**
In Cisco Unified CM Administration, choose **Media Resources > Music On Hold Audio Source**. The **Find and List Music On Hold Audio Sources** window appears.

**Step 2**
Enter search criteria to update an existing audio source. To list all records in the database, ensure that the dialog box is empty. Click **Find**.

**Step 3**
Click **Add New** to add a new Music On Hold audio source.

**Step 4**
Configure the fields in the **Music On Hold Audio Source Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

**Step 5**
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.

**What to Do Next**

- (Optional) **Configure Fixed Music On Hold Audio Source**, on page 553
- **Configure Media Resource Group**, on page 555

**Related Topics**

**Audio Source Fields for Music On Hold**, on page 550

---

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Music On Hold Audio Source Information</strong></td>
<td></td>
</tr>
<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>
</tbody>
</table>
### Field Description

<table>
<thead>
<tr>
<th>Allow Multicasting</th>
<th>Check this check box to specify that the selected MOH audio source allows multicasting.</th>
</tr>
</thead>
</table>
| MOH Audio Source File Status | This pane displays the following information about the source file for the selected MOH audio source:  
  - InputFileName  
  - ErrorCode  
  - ErrorText  
  - DurationSeconds  
  - DiskSpaceKB  
  - LowDateTime  
  - HighDateTime  
  - OutputFileList  
  - MOH Audio Translation completion date  
  Note: OutputFileList includes information on ULAW, ALAW, G.729, and Wideband wav files and status options. |

### Announcement Settings

- **Initial Announcement**  
  Choose an initial announcement from the drop-down list.  
  Note: To select MoH with no initial announcement, choose the **Not Selected** option.  
  Click the **View Details** link to view the following Initial Announcement information:  
  - Announcement Identifier  
  - Description  
  - Default Announcement  
  Note: Played by MOH server only when the Audio Source has “Allow Multi-casting” unchecked and “Initial Announcement Played” set to ‘Only for queued calls’.  
  Played by ANN if “Allow Multi-casting” is checked or if “Initial Announcement Played” is set to ‘Always.’

- **Initial Announcement Played**  
  Choose one of the following to determine when to play the initial announcement:  
  - Play announcement before routing to Hunt Member  
  - Play announcement if call is queued
### Field | Description
---|---
Periodic Announcement | Choose a periodic announcement from the drop-down list. **Note** To select MoH with no periodic announcement, choose the **Not Selected** option. Click the View Details link to view the following Periodic Announcement information:  
- Announcement Identifier
- Description
- Default Announcement
**Note** The MOH server always plays the periodic announcement regardless of other settings. Initial announcements are always simulcast to each new caller. Periodic announcements are multicast to queued callers at the specified time interval. Callers who join the queue after the periodic announcement has begun to play may only hear a portion of the announcement.

Periodic Announcement Interval | Enter a value (in seconds) that specifies the periodic announcement interval. Valid values are 10 to 300. The default value is 30.

Locale Announcement | Locale Announcement depends upon the locale installation package that has been installed.  
**Note**  
- Prompts played by MOH will use the setting for Locale Announcement.
- Prompts played by ANN will use the User Locale of the calling party.

**MoH Audio Sources**

(list of MoH audio sources) | 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 Cisco Unified Communications Manager server and performs audio conversions to create codec-specific audio files for MOH. Depending on the size of the original file, processing may take several minutes to complete.

**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 Fixed Music On Hold Audio Source

The Music On Hold server supports one fixed-device stream source in addition to the file stream sources. This source represents the fixed audio source, which you configure in the **Fixed MOH Audio Source Configuration** window. The fixed audio source originates from a fixed device that uses the local computer audio driver.

For each cluster, you may define one fixed audio source. You must set up the fixed audio source that is configured per cluster on each MOH server. To do so, connect a Cisco USB MOH sound adapter (which must be ordered separately) into the USB port for each MOH server in the cluster that you want to provide the fixed audio source.

**Note**
For virtual servers, 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, internal Music On Hold is supported on VMware.

### Before You Begin

Configure Music On Hold Audio. Perform the following procedures:

- [Upload Music On Hold Audio File](#), page 548
- [Convert Music On Hold Files](#), page 549
- [Configure Music On Hold Audio Source](#), page 549
Procedure

Step 1  In the Cisco Unified CM Administration, choose Media Resources > Fixed MOH Audio Source. The Fixed MOH Audio Source Configuration window appears.

Step 2  Configure the fields in the Fixed MOH Audio Source Configuration window. See the Related Topics section for more information about the fields and their configuration options.

Step 3  Click Save.

What to Do Next
Configure Media Resource Group, on page 555

Related Topics
Fixed Music on Hold Audio Source Fields for Music On Hold, on page 554

Fixed Music on Hold Audio Source Fields for Music On Hold

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Fixed MOH Audio Source Information</strong></td>
<td></td>
</tr>
<tr>
<td>Source ID</td>
<td>This field displays the stream number for this fixed MOH audio source.</td>
</tr>
<tr>
<td>Name</td>
<td>Enter a unique name in this field for the fixed MOH audio source. This name can comprise up to 50 characters. Valid characters include letters, numbers, spaces, dashes, dots (periods), and underscores. <strong>Note</strong> For virtual servers, the fixed Music On Hold device cannot specify an audio source that connects through a Universal Serial Bus (USB), because Cisco Unified Communications Manager does not support USB when running on VMware. Internal Music On Hold is supported on VMware.</td>
</tr>
<tr>
<td>Allow Multi-casting</td>
<td>Check this check box to specify that this fixed MOH audio source allows multicasting.</td>
</tr>
<tr>
<td>Enable (If checked, Name is required.)</td>
<td>To enable this fixed MOH audio source, check this check box.</td>
</tr>
</tbody>
</table>

Announcement Settings for Held and Hunt Pilot Calls
### Field | Description
--- | ---
Initial Announcement | Choose an initial announcement from the drop-down list box. **Note** To select MOH with no initial announcement, choose the default option, which is Not Selected. Select View Details to view the following Initial Announcement information:
- Announcement Identifier
- Description
- Default Announcement
**Note** To disable Initial Announcement completely, set Initial Announcement to Not Selected and set Initial Announcement Played to Only for Queued Calls.

Initial Announcement for queuing-enabled Hunt Pilot calls | Choose one of the following options from the drop-down list.
- Play announcement before routing to Hunt Member
- Play announcement if call is queued

Periodic Announcement | Choose a periodic announcement from the drop-down list: **Note** To select MOH with no periodic announcement, choose the default option, which is Not Selected. Click the View Details link to view the following Periodic Announcement information:
- Announcement Identifier
- Description
- Default Announcement

Periodic Announcement Interval | Enter a value (in seconds) that specifies the periodic announcement interval. Valid values specify 10 to 300. The default value is 30.

Locale Announcement | Locale Announcement depends upon the locale installation package that has been installed.

---

**Configure Media Resource Group**

Media Resource Group is a logical grouping of media servers. 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.

**Before You Begin**

Configure Music On Hold Audio. Perform the following procedures:
Procedure

Step 1  In the Cisco Unified CM Administration, choose Media Resources > Media Resource Group.

Step 2  Click Add New if you have to configure a new Media Resource Group.
     The Media Resource Group Configuration window appears.

Step 3  Enter search parameters to find a Media Resource Group if you have to configure an existing Media Resource Group.
     The system displays the records that match all the criteria in the Media Resource Group Configuration window.

Step 4  Configure the following fields in the Media Resource Group Configuration window:
     • Name—Enter a name for the media resource group.
     • Available Media Resources—From this list, select one or multiple media resources.
     • Selected Media Resources—Using the arrow key, select one or multiple media resources to use for multicasting.

Step 5  Click Save.
     This media resource group is then configured to be a member of a Media Resource Group List (MRGL). The MRGL is associated with devices, such as phones.

What to Do Next
Configure Media Resource Group List, on page 556

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.

• Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a Music On Hold resource.

• Following are the two levels of prioritized media resource group list selection:
  ° Level two media resource group list—Provides the higher priority level, which is device based. Cisco Unified Communications Manager uses the media resource group list at the device level if this media resource group list is defined.
  ° Level one media resource group list—Provides the lower priority level, which is an optional DevicePool parameter. Cisco Unified Communications Manager uses the DevicePool level media resource group list only if no media resource group list is defined in the device level for that device.
• If no media resource group lists are defined, Cisco Unified Communications Manager uses the system default resources. System default resources comprise resources that are not assigned to any existing media resource group. Ensure that system default resources are unicast.

Before You Begin
Configure Media Resource Group, on page 555

Procedure

Step 1 In Cisco Unified CM Administration, choose Media Resources > Media Resource Group List.

Step 2 Click Add New if you have to configure a new Media Resource Group List. The Media Resource Group List Configuration window appears.

Step 3 Enter search parameters to find a Media Resource Group List if you have to configure an existing Media Resource Group List. The system displays the records that match all the criteria.

Step 4 Configure the following fields in the Media Resource Group List Configuration window:

- Name—Enter a name for the media resource group list.
- Available Media Resource Groups—From this list, select one or multiple media resource groups.
- Selected Media Resource Groups—Using the arrow key, select one or multiple media resource groups.

Step 5 Click Save.

What to Do Next
View Music on Hold Audio File, on page 557

View Music on Hold Audio File

Perform the following procedure to view music on hold audio files that are stored on the system.

Before You Begin
Configure Media Resource Group List, on page 556

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.

---

**What to Do Next**

Enable Security for Music On Hold, on page 558

---

**Enable Security for Music On Hold**

You can enable the security mode of an MOH server in a cluster. Select a value in the **Cluster Security Mode** enterprise parameter so that the Music On Hold devices are automatically enabled for security. Enter one of the following values for this parameter:

- **0**—Implies Non Secure, which means that the cluster allows the phones to register with no security.
- **1**—Implies Mixed, which means that the cluster allows the registration of both secure devices and non-secure devices.

**Note**

The **Cluster Security Mode** enterprise parameter is a non-editable parameter. To change the cluster security mode, you must run the Certificate Trust List (CTL) Client plugin. Then, you must restart Cisco Unified Communications Manager for the parameter change to take effect.

---

**Before You Begin**

View Music on Hold Audio File, on page 557

---

**Procedure**

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

**Step 2** In the **Security Parameters** section, set the **Cluster Security Mode** option to 1.

---

**What to Do Next**

(Optional) Enable Secured Music On Hold through SRTP, on page 559

Configure multicast by performing the following subtasks:

- Plan Music On Hold Server Capacity, on page 561
Enable Secured Music On Hold through 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.

In a secure mode, the Cisco Unified Communications Manager Administration device page for Music On Hold displays a Device is trusted message with a green check box, indicating that it is a trusted device.

Note

Before You Begin

Enable Security for Music On Hold, on page 558

Procedure

Step 1
In the Cisco Unified CM Administration, choose System > Enterprise Parameters.

Step 2
In Enterprise Parameters window, set the Cluster Security Mode parameter to Mixed Mode.

Note
The media streaming between the devices is done through SRTP. When calls are secure, an icon with a secured lock appears on the Cisco Unified IP Phone, indicating that the call is protected for both signaling and media.

This parameter indicates the security mode of the cluster. A value of 0 indicates Non Secure (phones register in nonsecure mode); 1 indicates Mixed (the cluster allows the registration of both secure devices and nonsecure devices). Because this parameter is read-only, to change the cluster security mode, you must run the CTL Client plugin.

Step 3
Click Save, and restart Cisco Unified Communications Manager for the parameter change to take effect.

What to Do Next

Configure multicast by performing the following subtasks:

• Plan Music On Hold Server Capacity, on page 561
• Verify Music On Hold Service Parameters, on page 562
• Configure Multicast Music On Hold Audio Sources/Fixed MOH Audio Source, on page 563
• Configure Multicast Music On Hold Server, on page 563
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 16: 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>
<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>
</tbody>
</table>
### Multicast Configuration

**Plan Music On Hold Server Capacity**

It is crucial to plan the capacity of the deployed and configured hardware and to ensure the support it can provide for 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.

**Before You Begin**
- Enable Security for Music On Hold, on page 558
- (Optional) Enable Secured Music On Hold through SRTP, on page 559

**Procedure**

**Step 1** In Cisco Unified CM Administration, choose Media Resources > Music on Hold Server.

**Step 2** In the Find and List Music On Hold Servers window, enter the search parameters and click Find. The system displays the records that match all the criteria.

**Step 3** Check the check box of the MOH Server for which you want to plan the capacity. The Music On Hold (MOH) Server Configuration window appears.

**Step 4** Configure the following fields in the Music On Hold (MOH) Server Configuration window:

<table>
<thead>
<tr>
<th><strong>Unicast Audio Source</strong></th>
<th><strong>Multicast Audio Source</strong></th>
</tr>
</thead>
<tbody>
<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 Use Multicast for MOH Audio 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, 7961, 7970, and 7971, set this parameter to sendOnly. Otherwise, leave this parameter set to the default value, recvOnly.
• **Maximum Half Duplex Streams**—This parameter determines the number of devices that you can place on unicast MOH. By default, this value is set to 250. Set this parameter to the value that is derived from the following formula:

\[
\text{(Server and deployment capacity)} - \left(\left[\text{Number of multicast MOH sources}\right] \times \left[\text{Number of enabled MOH codecs}\right]\right)
\]

The value of this parameter should be set according to the platform and deployment type (coresident or standalone).

**Note**  Regarding the maximum suggested number of MOH streams (250 MOH streams on Cisco MCS 7815 and 7825 Series and 500 MOH streams on Cisco MCS 7835 and 7845 Series) – Count each multicast audio source as two MOH streams. For example, for Cisco MCS 7835 and 7845 Series, if three multicast MOH audio sources and four codecs are enabled, no more than 476 unicast MOH streams should be generated at the same time (\(2 \times 3 \times 4 + 476 = 500\)).

• **Maximum Multi-cast Connections**—This parameter determines the number of devices that you can place on multicast MOH. By default, this value is set to 30,000. Set this parameter to a value that ensures that all devices can be placed on multicast MOH, if necessary. Although the MOH server can generate only a finite number of multicast streams (a maximum of 204), many held devices can join each multicast stream. This parameter should be set to a number that is greater than or equal to the number of devices that might be placed on multicast MOH at any given time.

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

The changes take place when the streaming to the device is idle.

---

**What to Do Next**

[Verify Music On Hold Service Parameters, on page 562](#)

**Verify Music On Hold Service Parameters**

Perform the following procedure to verify the Music On Hold server and its service parameters:

**Before You Begin**

[Plan Music On Hold Server Capacity, on page 561](#)

**Procedure**

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

The **Service Parameter Configuration** window appears.

**Step 2**  Select a server from the **Server** drop-down list.

After you select a server, the **Service** field appears.

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

The server and service parameters appear in the **Service Parameter Configuration** window.

**Step 4**  Verify the server and service parameters.

**Note**  All the parameters apply only to the current server except the parameters that are in the cluster-wide groups.

**Step 5**  Click **Save**.
What to Do Next
Configure Multicast Music On Hold Audio Sources/Fixed MOH Audio Source, on page 563

Configure Multicast Music On Hold Audio Sources/Fixed MOH Audio Source
For multicast to be available, configure the Cisco Unified Communications Manager services to allow multicasting on MOH audio sources or fixed MOH audio source.

Before You Begin
Verify Music On Hold Service Parameters, on page 562

Procedure

Step 1  In Cisco Unified CM Administration, choose Media Resources > Music On Hold Audio Source.
Step 2  Enter search parameters to find a Music On Hold audio source. The system displays the records that match all the criteria.
Step 3  In the Music On Hold Audio Source Configuration window, check the Allow Multi-casting checkbox to allow multicasting.
Step 4  Click Save.

What to Do Next
Configure Multicast Music On Hold Server, on page 563

Configure Multicast Music On Hold Server
After you allow multicast Music On Hold (MOH) on audio sources, you must enable the MOH server for multicast Music on Hold.

When you use multicast MOH and when 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 multicasts on interfaces on which the multicast MOH packets are not required and use the Max Hops parameter.

Note
To use multicast MOH when you use Media Resource Group and Media Resource Group Lists to implement media-resources access control and when you assign a multicast MOH server to a Media Resource Group, you must also enable multicast MOH for the Media Resource Group.

Before You Begin
Configure Multicast Music On Hold Audio Sources/Fixed MOH Audio Source, on page 563
Procedure

Step 1 In Cisco Unified CM Administration, choose Media Resources > Music On Hold Server.

Step 2 Enter search parameters to find a Music On Hold server.

Step 3 In the Music On Hold (MOH) Server Configuration window, check the Enable Multi-cast Audio Sources on this MOH Server checkbox. The Base Multi-cast IP Address, Base Multi-cast Port Number, and Increment Multi-cast On fields are populated automatically. You can modify these values as desired.

Step 4 (Optional) Configure the following fields in the Music On Hold (MOH) Server Configuration window:

- **Base Multi-cast IP Address**—Enter the multicast IP addresses that range from 224.0.1.0 to 239.255.255.255.
  
  **Note** IP addresses between 224.0.1.0 and 238.255.255.255 fall in the reserved range of IP multicast addresses for public multicast applications. Use of such addresses may interfere with existing multicast applications on the Internet. Use IP addresses in the range that is reserved for administratively controlled applications on private networks (239.0.0.0 - 239.255.255.255).

- **Base Multi-cast Port Number**—Enter the multicast port numbers that include even numbers and range from 16384 to 32766.
  
  **Note** Increment multicast on IP address instead of on port number. Doing so results in each multicast audio source to have a unique IP address and helps to avoid network saturation in firewall situations.

- **Increment Multi-cast On**—Click Port Number to increment multicast on port number or click IP Address to increment multicast on IP address.
  
  **Note** All MOH audio sources that you configure to allow multicasting are listed in the Selected Multicast Audio Sources section of the Music On Hold (MOH) Server Configuration window.

Step 5 Click Save.

---

**What to Do Next**

Configure a Multicast-Enabled Media Resource Group, on page 564

**Configure a Multicast-Enabled Media Resource Group**

Multicast Music On Hold (MOH) works only if you assign the Multicast-enabled MOH server to a Multicast-enabled Media Resource Group. Configure this Media Resource Group to be a member of a Media Resource Group List. Then, you can associate the Media Resource Group List with devices, such as phones.

**Before You Begin**

- Assign a multicast-enabled MOH server to a multicast-enabled Media Resource Group for the multicast MOH to work.

- Configure Multicast Music On Hold Server, on page 563
### Procedure

**Step 1** In Cisco Unified CM Administration, choose **Media Resources > Media Resource Group**.

**Step 2** Enter search parameters to find a Media Resource Group. The system displays the records that match all the criteria.

**Step 3** Configure the following fields in the **Media Resource Group Configuration** window:

- **Name**—Enter a name for the media resource group
- **Available Media Resources**—From this list, select one or multiple media resources.
- **Selected Media Resources**—Using the arrow key, select one or multiple media resources to use for multicasting.

**Step 4** Check the **Use Multi-cast for MOH Audio** check box, if at least one multicast resource is available.

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

This media resource group is then configured to be a member of a Media Resource Group List (MRGL). The MRGL is associated with devices, such as phones.

### What to Do Next

**Configure Multicast Music On Hold over H.323 Intercluster Trunks, on page 565**

#### Configure 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).

Consider these guidelines for configuring multicast MOH:

- 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 remains active by default. To turn off the feature, set the value of 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.

### Procedure

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

**Step 2** In the **Service Parameter Configuration** window, select a server and the CallManager service.

**Step 3** In the **Clusterwide Parameters (Service)** section, set the value of **Send Multicast MOH in H.245 OLC Message** service parameter to **True**.

**Step 4** Click **Save**.
Reset or Restart a Music On Hold Server

Perform the following procedure to reset or restart an existing Music On Hold server.

**Before You Begin**

Configure multicast by performing the following subtasks:

- Plan Music On Hold Server Capacity, on page 561
- Verify Music On Hold Service Parameters, on page 562
- Configure Multicast Music On Hold Audio Sources/Fixed MOH Audio Source, on page 563
- Configure Multicast Music On Hold Server, on page 563
- Configure a Multicast-Enabled Media Resource Group, on page 564
- Configure Multicast Music On Hold over H.323 Intercluster Trunks, on page 565

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose Media Resources &gt; Music On Hold Server.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Check the check box for the Music On Hold server that you want to reset, and click Reset. A popup window shows an information message.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Click Restart to restart the Music On Hold server, or click Reset to reset the Music On Hold server.</td>
</tr>
</tbody>
</table>

**What to Do Next**

(Optional) Synchronize Music On Hold Server, on page 566

Synchronize Music On Hold Server

To synchronize a Music on Hold Server with the most recent configuration changes, perform the following procedure. After you perform this procedure any outstanding configuration is applied in the least-intrusive manner possible. For example, a reset or restart may not be required on few affected devices.

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>In Cisco Unified CM Administration, choose Media Resources &gt; Music On Hold Server.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Enter search parameters to find a Music On Hold server, and click Find.</td>
</tr>
</tbody>
</table>
The system displays the search results for the records that match all the criteria.

**Step 3**
Check the check boxes next to the Music On Hold servers that you want to synchronize. To select all MOH servers in the window, check the check box in the matching records title bar.

**Step 4**
Click **Apply Config to Selected**.

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

---

**Music on Hold Interactions and Restrictions**

**Music On Hold Interactions**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multicast Music On Hold over H.323 Intercluster Trunks</td>
<td>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.</td>
</tr>
<tr>
<td>Music On Hold Failover and Fallback</td>
<td>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.</td>
</tr>
</tbody>
</table>
Music On Hold Interactions and Restrictions

<table>
<thead>
<tr>
<th>Feature</th>
<th>Interaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Park and Directed Call Park</td>
<td>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:</td>
</tr>
<tr>
<td></td>
<td>• User hold—The system invokes this type of hold when a user presses the Hold button or Hold softkey.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td>Extension Mobility Cross Cluster—Media resources for the visiting phone</td>
<td>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).</td>
</tr>
<tr>
<td>Hold Reversion</td>
<td>Cisco Unified Communications Manager supports MOH on a reverted call if MOH is configured for a normal held call.</td>
</tr>
</tbody>
</table>

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><strong>Restriction</strong></td>
<td><strong>Description</strong></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, the caller receives silence instead of Music On Hold. To avoid this situation, configure unicast MOH or Tone on Hold 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>Restriction</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</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>MOH Audio Source Identifier</td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td>Administrators for Consistent Caller-specific MOH Configurations</td>
<td>Administrators are responsible to maintain consistent caller-specific MOH configurations when multiple Cisco Unified Communications Manager clusters are involved.</td>
</tr>
<tr>
<td>Original Incoming Caller</td>
<td>The original incoming caller to the call center cannot change during the course of the entire call.</td>
</tr>
<tr>
<td>MOH Information</td>
<td>The Music On Hold information is shared only across SIP trunks.</td>
</tr>
</tbody>
</table>

**Troubleshooting Music On Hold**

**Music On Hold Does Not Play on Phone**

**Problem**

Phone user cannot hear Music On Hold.

**Possible Cause**

- 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.

**Solution**

- Verify the IP addressing mode of the device where Music On Hold is played. If the IP addressing mode for the device is IPv6 Only and if Music On Hold is configured for unicast Music On Hold, ensure that a dual-stack MTP is configured and available for media translation.
• 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 and control phone features and settings.

As the administrator, you control access to the Self Care Portal. You must also provide information to your users so that they can access the Self Care Portal.

Before a user can access the Cisco Unified Communications Self Care Portal, you must add the user to a standard Cisco Unified Communications Manager End User Group.

You must provide users with the following information:

- Use one of the following URLs to access the application:
  - 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.

- A user ID and default password to access the application.
- An overview of the tasks that users can accomplish with the portal.

These settings correspond to the values that you enter when you add the user to Cisco Unified Communications Manager End User Group.
Self Care Portal Task Flow

Procedure

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> Grant Access to the Self Care Portal, on page 574</td>
<td>Use this procedure to enable a user to access the Self Care Portal.</td>
</tr>
<tr>
<td><strong>Step 2</strong> Configure the Self Care Portal Display, on page 574</td>
<td>Use this procedure to choose the Self Care Portal default server and to choose which features and settings are enabled on the Self Care Portal display.</td>
</tr>
</tbody>
</table>

Grant Access to the Self Care Portal

Use this procedure to enable a user to access the Self Care Portal.

Procedure

1. From Cisco Unified Communications Manager Administration, select **User Management > End User**.
2. Search for the user for whom you want to provide access to the Self Care Portal and click the user ID link.
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.
4. In the **Permission Information** section, click **Add to Access Control Group**.
5. From the list, choose **Standard CCM End Users** to add to the **Groups** box.
6. Select **Save**.

Configure the Self Care Portal Display

The Self Care Portal parameters that you enable or disable apply to all of the Self Care Portal pages on the Self Care Portal server.

Procedure

1. From Cisco Unified Communications Manager Administration, select **System > Enterprise Parameters**.
2. In the Self Care Portal area, 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.
3. Enable or disable the parameters that the users can access in the portal.
For descriptions of the parameters, see the Enterprise Parameters section in the Online Help.

**Step 4** Select **Save**.