

Feature Configuration Guide for Cisco Unified Communications Manager, Release 15 and SUs

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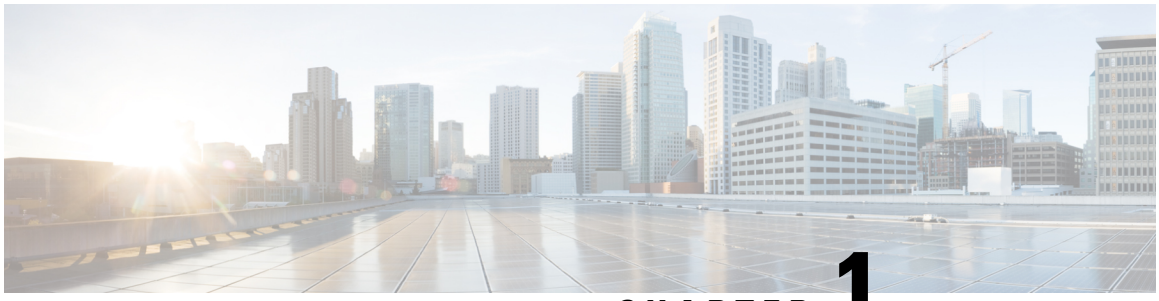
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CHAPTER 1

New and Changed Information

- New and Changed Information, on page 1

New and Changed Information

The following table provides an overview of the significant changes to the features in this guide up to this current release. The table does not provide an exhaustive list of all changes made to the guide or of the new features up to this release.

Table 1: New Features and Changed Behavior in Unified Communications Manager and IM and Presence Service

Date	Description	See
July 31, 2025	• Removed support for Call Control Discovery via Service Advertisement Framework in Unified Communications Manager • Support for ringback and precedence call display for MLPP call from SIP endpoint	• Deprecation of Call Control Discovery via Service Advertisement Framework in Cisco Unified Communications Manager • Multilevel Precedence and Preemption Restrictions, on page 842
October 01, 2024	Support for TLS 1.3	• Configure Redirector Servlet, on page 278 • Configure Connection to InformaCast, on page 305 • Extension Mobility Cross Cluster Overview, on page 421
October 01, 2024	Added prerequisites for Windows 11 support	Manager Assistant Prerequisites, on page 161

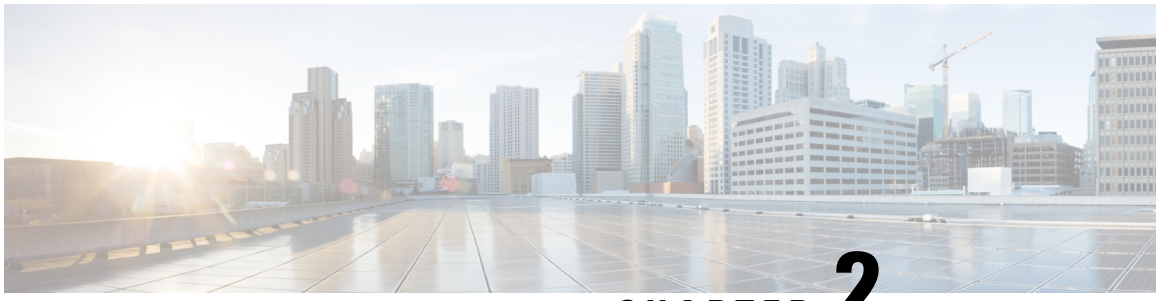
Date	Description	See
December 18, 2023	Windows 11 support for Unified Communications Manager Assistant Administration and the Assistant Console.	Manager Assistant Prerequisites, on page 161



PART I

Getting Started

- Feature Configuration Overview, on page 5
- Configuration Tools, on page 7



CHAPTER 2

Feature Configuration Overview

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

About the Feature Configuration Guide

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

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, choose **System Reports**.

Step 2 From the list of reports, click **Unified CM Phone Feature List**.

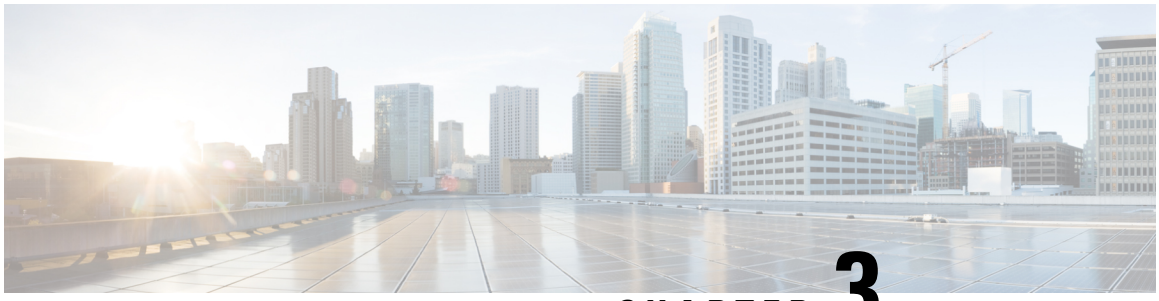
Step 3 Perform one of the following steps:

- Choose **Generate New Report** (the bar chart icon) to generate a new report.
- Choose **Unified CM Phone Feature List** if a report exists.

Step 4 From the **Product** drop-down list, choose **All**.

Step 5 Click the name of the feature that you want to configure.

Step 6 Click **Submit**, to generate the report.



CHAPTER 3

Configuration Tools

- About the Feature Configuration Guide, on page 7
- Configuration Tools Overview, on page 7
- Generate a Phone Feature List, on page 9

About the Feature Configuration Guide

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

Configuration Tools Overview

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

- Cisco Unified Communications Manager Administration
- Cisco Unified Serviceability

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

Cisco Unified Communications Manager Administration

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

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

Log In to Cisco Unified CM Administration

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

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

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

Procedure

Step 1 Start your preferred operating system browser.

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

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

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

Note
You can optionally specify a port number.

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

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

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

Cisco Unified Communications Manager Serviceability

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

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

- Saves alarms and events for troubleshooting and provides alarm message definitions.
- Saves trace information to log files for troubleshooting.

- Monitors real-time behavior of components through the Cisco Unified Real-Time Monitoring Tool (Unified RTMT).
- Provides audit capability by logging configuration changes to the system by a user or due to result of the user action. This functionality supports the Information Assurance feature of Unified Communications Manager and Cisco Unity Connection.
- Provides feature services that you can activate, deactivate, and view through the **Service Activation** window.
- Generates and archives daily reports; for example, alert summary or server statistic reports.
- Allows Unified Communications Manager, 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.
- **Unified Communications Manager only**: Generates Unified Communications Manager reports for Quality of Service, traffic, and billing information through Cisco Unified Communications Manager CDR Analysis and Reporting.

Log into Cisco Unified Communications Manager Serviceability

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

Procedure

Step 1 Start your preferred operating system browser.

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

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

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

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

Step 4 From Cisco Unified CM Administration, choose **Cisco Unified Serviceability** from the Navigation menu drop-down list and click **Go**.

Step 5 Enter the username and password that you specified during Unified Communications Manager installation and click **Login**.

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

Generate a Phone Feature List

Generate a phone feature list report to determine which devices support the feature that you want to configure.

Procedure

Step 1 From Cisco Unified Reporting, choose **System Reports**.

Step 2 From the list of reports, click **Unified CM Phone Feature List**.

Step 3 Perform one of the following steps:

- Choose **Generate New Report** (the bar chart icon) to generate a new report.
- Choose **Unified CM Phone Feature List** if a report exists.

Step 4 From the **Product** drop-down list, choose **All**.

Step 5 Click the name of the feature that you want to configure.

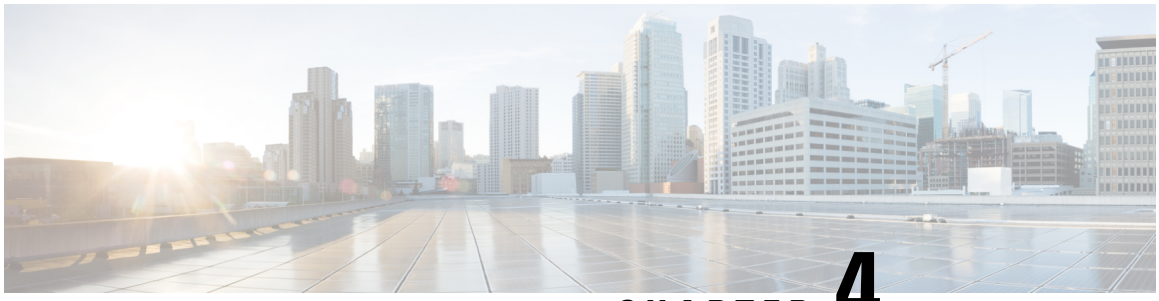
Step 6 Click **Submit**, to generate the report.



PART II

Remote Worker Features

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CHAPTER 4

Cisco Unified Mobility

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Cisco Unified Mobility Overview

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

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

Benefits of Cisco Unified Mobility

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

This provides the following benefits:

- Single enterprise phone number and voicemail for all business calls, regardless of which device you are using, and whether you are in the office or out of the office.
- Ability to extend business calls to a mobile device and have the call still be handled as if it were your office phone.
- Calls placed from mobile devices are anchored to the enterprise and routed through an enterprise gateway. This provides access to UC mid-call features, centralized billing and call detail records, and potential cost savings from avoiding expensive cellular networks.

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

Wi-Fi to LTE Call Handoff

Important This section is applicable from Release 14SU1 onwards.

This feature provides flexibility to the soft client end users to switch between Wi-Fi and LTE networks or vice versa without disconnecting any active calls while switching networks. Wi-Fi to LTE Call Handoff feature is automatically enabled but requires Unified Communications Manager release 14SU1 and later.

During the call, when the soft client detects the change in the network, switches registration, and reconnects the active call with an audio-visual indication to the end user about the switch. However, the users continue to have seamless audio and video experience on the call.

Note This feature supports only the active call handover. If Call Recording is active, the recording is stopped and does not continue after handover. Also, the network handover does not support the mid-call features (such as hold or transfer), screen share, conference call, and call center features. For more information, see the '*Prepare Your Environment for Calling in Webex (Unified CM)*' chapter in Deployment Guide for Calling in Webex (Unified CM).

Cisco Desktop and the latest Webex Mobile (WebexApp 41.8) versions support this feature. For more information, see the '*Known Issues and Limitations with Calling in Webex (Unified CM)*' section in Deployment Guide for Calling in Webex (Unified CM).

Mobility Features

Cisco Unified Mobility offers the following mobility-related features:

Mobility Feature	Description
Single Number Reach	Provides you with a single enterprise phone number and voicemail by which a caller can reach you, regardless of whether you are in the office or outside the office. When someone dials your enterprise number, you can answer the call from your desk phone, or from any of your configured remote destinations (for example, a home office phone, a dual-mode Cisco Jabber on iPhone or Android client, and even a mobile phone from another provider) .

Mobility Feature	Description
Move to Mobile	Allows you to transfer an active call from your desk-phone to a mobile device that is configured as a remote destination by pressing the **Mobility** softkey on your Cisco IP Phone. It is associated with Single Number Reach as a part of the Remote Destination configuration. Similar to the **Move to Mobile** option is the **Desk Pickup** option, which fits the example where you are on a mobile call and are just arriving at the office. You can hang up on the call on your mobile device and immediately resume the call by picking up your desk phone before the **Maximum Wait Time for Desk Pickup** timer expires (the default is 10 seconds). This option is enabled as part of your Single Number Reach configuration. • Ensure that you set the **Enforce Privacy Setting on Held Calls** Service Parameter to False. • You can also use the `Enterprise Feature Access` code and the `Session Handoff` codes to transfer calls between your remote destinations and desk phone.
Mobile Voice Access	Allows you to place calls from any remote phone and have the call be anchored in the enterprise and presented to the called party as if you had called from your office phone. When using this feature, you must dial in to a system interactive voice response from your mobile device. After authenticating you, and prompting you for the call destination, the system places the call as if you had called from your enterprise phone. You can also use **Mobile Voice Access** prompts to enable or disable **Single Number Reach** for a remote destination.
Enterprise Feature Access .	Provides two-stage dialing from a configured remote destination. Also, ensures that the call that is presented to the called party appears as if it originated from your desk phone. Unlike **Mobile Voice Access**, to use **Enterprise Feature Access**, you must be dialing from one of your configured remote destinations. **Enterprise Feature Access** also allows you to access mid-call features while on a call from a remote destination. You can access mid-call features by sending DTMF digits that represent the codes for the various features such as Hold, Exclusive Hold, Transfer.
Intelligent Session Control	Enables automatic call anchoring for enterprise-originated calls that are placed directly to configured remote destination numbers (for example, an enterprise-originated call to a cell phone number that is configured as a remote destination). By configuring a service parameter, you can have the system redirect those calls automatically to the associated enterprise number, providing cost savings and added UC functionality.

Mobility Feature	Description
Dual-Mode Phones	Cisco Jabber on iPhone and Android clients can be provisioned as dual-mode devices. **Dual-Mode phones** have the capability of connecting over Wi-Fi or through cellular networks. When the client is within the enterprise network, Cisco Jabber can register to Unified Communications Manager over Wi-Fi, and has UC calling and instant messaging functionality. If you configure a mobile identity with the phone number of the mobile device, allowing the call to be transferred from Jabber to the cellular device when leaving the enterprise network. **Note** An added feature that is available to Cisco Jabber mobile clients is Mobile and Remote Access, which allows Cisco Jabber clients to connect to data networks when outside of the enterprise network. For more information, see "*Configure Mobile and Remote Access*" section in Feature Configuration Guide for Cisco Unified Communications Manager.

Cisco Unified Mobility Prerequisites

Refer to the following prerequisites:

- Enabling Mobility features requires proper planning to ensure that your dial plan and call routing configuration can handle the deployment needs. For more information, see "*Mobile Collaboration*" section in the *Cisco Collaboration System Solution Reference Network Designs* guide.
- For information on which Cisco IP Phones support Mobility feature, see Generate a Phone Feature List, on page 5.
  - For a list of Cisco IP Phones that support the Mobility softkey, run a report for the **Mobility** feature.
  - For a list of supported dual-mode phones, run a report for the **Dual-Mode** feature.
- If you are deploying Mobile Voice Access and you want to make additional locales available to your system (if you want to use non-English phone locales or country-specific tones), you can download the locale installers from cisco.com and install them through the Cisco Unified OS Administration interface. For more information on installing locales, see Installation Guide for Cisco Unified Communications Manager and the IM and Presence Service.
- Configure Self-Provisioning so that phone users can provision their own Cisco Jabber clients and remote destinations. For more information, see "*Configure Self Provisioning*" and "*Provisioning End Users*" section in the System Configuration Guide for Cisco Unified Communications Manager.

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 Configuration Task Flow

Complete these tasks to configure Mobility features for your deployment.

Procedure

	Command or Action	Purpose
Step 1	Perform one of the following: • Configure a Mobility User, on page 18 • Configure Mobility Users Through Bulk Administration, on page 18 • Provision Mobility Users Through LDAP, on page 19	Adds mobility features for an individual end user. Configures Mobility features for a large number of existing end users, use the Bulk Administration Tool. Provisions new users with mobility functionality, you can use a feature group template and LDAP sync.
Step 2	Configure Mobility for IP Phones, on page 20	Configures Cisco IP Phones for Mobility including setting up the Single Number Reach (SNR) and Move to Mobile features. This allows enterprise phone users to extend enterprise calls to a wide range of mobile devices, including a home office phone or a mobile phone.
Step 3	Configure Mobile Voice Access, on page 25	**Optional**. Provides a system IVR so that mobile users can call from any mobile device and have the call that is presented to the called party as if the caller were dialing from their enterprise desk phone.
Step 4	Configure Enterprise Feature Access, on page 27	**Optional**. Provides two-stage dialing from a configured remote destination and have the call that is presented to the called party as if it originated from a desk phone. This feature also allows you to access mid-call features while on a call from a remote destination.
Step 5	Configure Intelligent Session Control, on page 28	Configure the system so that inbound calls to a remote destination are rerouted to an associated enterprise, if one is available. This provides automatic call anchoring within the enterprise for mobility calls, providing cost savings and added Unified Communications functionality.
Step 6	Configure Mobility Service Parameters, on page 29	**Optional**. Configure optional mobility-related service parameters if you want to change the behavior of Cisco Unified Mobility.

	Command or Action	Purpose
Step 7	Configure Cisco Jabber Dual-Mode, on page 29	Configure Cisco Jabber for mobility so your users can access enterprise communications features through a Jabber client on their smartphone.
Step 8	Configure Other Dual-Mode Devices, on page 30	Complete this task flow if you want to deploy other dual-mode devices, such as FMC or IMS clients that can connect through Wi-Fi.

Configure a Mobility User

Use this procedure to configure an end user with the mobility feature.

Procedure

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

Step 2 In **Find and List Users** window, perform one of the following tasks:

- Click **Find** and select an existing user to modify the settings.
- Click **Add New** to configure a new user.

Step 3 Configure values for the following fields:

- **User ID**
- **Last Name**

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

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

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

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

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

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

Step 6 Click **Save**.

Configure Mobility Users Through Bulk Administration

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

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

Procedure

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

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

Step 3 Click **Next**.

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

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

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

Step 6 Click **Submit**.

Provision Mobility Users Through LDAP

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

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **User/Phone Add** > **Feature Group Template**.

Step 2 In the **Find and List Feature Group Templates** window, perform one of the following:

- Click **Add New** to configure a new template.
- Click **Find** and select an existing template to configure.

Step 3 Assign a **Name** to the template.

Step 4 Configure the following Mobility fields:

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

Step 5 Configure the remaining fields in the **Feature Group Template Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 6 Click **Save**.

Note
Assign the configured Feature Group Template to an LDAP Directory that has not yet been synced. Newly synced users have Mobility enabled. For more information, on provisioning users through LDAP see "*Provisioning End Users*" chapter in System Configuration Guide for Cisco Unified Communications Manager.

Configure Mobility for IP Phones

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

Procedure

	Command or Action	Purpose
Step 1	Configure Softkey Template for Mobility, on page 21	Configures a mobility softkey template for Cisco IP Phones that includes the Mobility softkey. Users can transfer calls from their deskphone to a mobile phone by pressing the softkey.
Step 2	Configure IP Phone for Mobility, on page 22	Configures an IP phone for mobility so that incoming calls to an enterprise number are extended to remote destinations.
Step 3	Configure a Remote Destination Profile, on page 23	Configures common settings that you want to apply to all the remote destination numbers for a user.

	Command or Action	Purpose
Step 4	Configure a Remote Destination, on page 24	Configures a remote destination that is a virtual device that represents a mobile device where the user can be reached (for example, a home office phone, or a mobile phone on a cellular network). The remote destination carries many of the same settings as the user's desk phone.
Step 5	Configure an Access List, on page 25	**Optional**. Controls which calls can ring which remote destinations, and at which times of day. The access list filters callers based on the Caller ID and can either allow calls or block calls from the caller during that remote destination's ring schedule.

Configure Softkey Template for Mobility

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

Procedure

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

Step 2 To create a new softkey template do the following. Otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) In the **Softkey Template Name** field, enter a new name for the template.
d) Click **Save**.

Step 3 To add mobility softkeys to an existing template.

a) Enter search criteria and click **Find**.
b) Choose an existing template.

Step 4 (Optional) Check the **Default Softkey Template** check box if you want to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Click **Save**.

Step 6 From the **Related Links** drop-down list, choose **Configure Softkey Layout** and click **Go**.

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

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

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

Step 10 Click **Save**.

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

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

Enable Mobility Within Feature Control Policy

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

Procedure

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

Step 2 Click **Find** and choose the applicable policy.

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

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

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

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

Step 6 Click **Save**.

Configure IP Phone for Mobility

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

Procedure

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

Step 2 Perform one of the following tasks:

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

Step 3 Click **Next**.

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

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

Note
You can configure either the **Owner User ID** or **Mobility User ID** field. Mobility users are configured for mobility-enabled devices and Owner users are configured for Non-Mobility devices. Configuring both users for the same device is not recommended.

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

Step 7 Click **Save**.

Configure a Remote Destination Profile

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

Procedure

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

Step 2 Click **Add New**.

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

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

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

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

Step 7 Click **Save**.

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

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

Configure a Remote Destination

A remote destination is a virtual device that represents a mobile device where the user can be reached (for example, a home office phone, a mobile phone on a cellular network, or a PSTN phone). The remote destination carries many of the same settings as the user's desk phone.

Note

- When an enterprise user initiates a call from a remote destination to Cisco Jabber, Unified Communications Manager tries to establish a data call with Cisco Jabber by sending an INVITE message to Cisco TelePresence Video Communication Server (VCS). The call is established regardless of receiving a response from VCS.
- If you have Self-Provisioning enabled, your end users can provision their own phones from the Self-Care Portal. See the System Configuration Guide for Cisco Unified Communications Manager and the "*Configure Self-Provisioning*" chapter for details on configuring the system for self-provisioning and the "*Provisioning End Users*" part for details on enabling self-provisioning for users as a part of a User Profile.

Procedure

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

Step 2 Click **Add New**.

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

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

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

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

Step 7 Use the **Single Number Reach Voicemail Policy** drop-down list to configure the voicemail policy.

a) Check the **Enable Single Number Reach** check box.

b) Check the **Enable Move to Mobile** check box to include this remote destination to the list of available destinations when the user presses the **Mobility** softkey on their desk phone.

Step 8 (Optional) If you want to limit enterprise calls to this remote destination to specific periods such as office hours, configure a **Ring Schedule**.

Step 9 In the **When receiving a call during the above ring schedule** area, apply the list that is configured for this remote destination.

Step 10 Configure the remaining fields on the **Remote Destination Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 11 Click **Save**.

Configure an Access List

An access list is an optional remote destination configuration if you want to control which calls can ring which remote destinations, and at which times of day. The access list filters callers based on the Caller ID and can either allow calls or block calls during that remote destination's ring schedule.

Procedure

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

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

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

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

Step 5 Choose one of the following options:

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

Step 6 Click **Save**.

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

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

Step 8 Choose **Save**.

Step 9 Apply the access list to a remote destination:

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

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

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

c) Click **Save**.

Configure Mobile Voice Access

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

Before you begin

To use Mobile Voice Access:

- Users must be enabled as mobility users with the **Enable Mobile Voice Access** option checked within **End User Configuration** For details, see Configure a Mobility User, on page 18.
- Interactive Voice Response service must be active, and included in a Media Resource Group List that the trunk uses.

Procedure

	Command or Action	Purpose
Step 1	Enable Mobile Voice Access, on page 26	Enable the Mobile Voice Access feature and specify a directory number that users can dial to reach the enterprise.
Step 2	Restart Cisco CallManager Service, on page 27	After you activate Mobile Voice Access, restart the Cisco CallManager service.

Note When the **Enforce End User PIN Lockout for Conference Now and Mobile Voice Access** service parameter is set to **True** (default), each failed Mobile Voice Access PIN attempt increments that user's credential hack count. Unified Communications Manager locks the PIN when the number of failures reaches the threshold configured in the assigned Credential Policy.

Mobile Voice Access and other PIN-authenticated features reject the locked PIN until an administrator unlocks the end user. Setting the parameter to **False** prevents subsequent Conference Now and Mobile Voice Access PIN failures from incrementing the hack count, but it does not unlock an end user who is already locked.

Enable Mobile Voice Access

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

Procedure

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

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

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

Step 4 Configure the following service parameters:

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

Step 5 Click **Save**.

Restart Cisco CallManager Service

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

Procedure

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

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

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

Step 4 Click **Restart**.

What to do next

You have now completed all the tasks that are required to configure Unified Communications Manager with native Mobile Voice Access support.

Configure Enterprise Feature Access

Use the following procedure to configure Enterprise Feature Access from a remote destination for:

- Two-stage dialing to place enterprise calls from a configured remote destination. Calls appear to the called party as if they were placed from an associated desk phone.
- Remote destination access to mid-call features through EFA codes that are sent using DTMF digits sent from the remote destination.

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

Procedure

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

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

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

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

Step 5 Click **Save**.

Step 6 Configure the Enteprise Feature Access service parameters:

a) From Cisco Unified CM Administration, choose **System** > **Service Parameters**.
b) From the **Server** drop-down list, choose the publisher node.
c) From the **Service** drop-down list, choose **Cisco CallManager**.
d) Set the **Enable Enterprise Feature Access** service parameter to **True**.
e) (Optional) In the **Clusterwide Parameters (System - Mobility)** area, edit the DTMF digits that you must enter to access midcall features through Enterprise Feature Access. For example, you could edit the **Enterprise Feature Access Code for Hold** service parameter, which has a default value of ***81**. The default values are as follows:

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

f) Click **Save**.

Configure Intelligent Session Control

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

Procedure

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

Step 2 From the **Server** drop-down list, choose a Cisco Unified Communications Manager node.

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

Step 4 Under **Clusterwide Parameters (Feature - Reroute Remote Desination Calls to Enterprise Number)** set the following service parameters:

- **Reroute Remote Destination Calls to Enterprise Number**—To enable Intelligent Session Control, set this parameter to **True**.

- **Ring All Share Lines**—Set the value of the parameter to **True**. If Intelligent Session Control is enabled, and this service parameter is also enabled, the system anchor calls to remote destinations within the enterprise, and will also ring all the user's shared lines.
- **Ignore Call Forward All on Enterprise DN**—This parameter applies only to outgoing calls to a remote destination when Intelligent Session Control is enabled. By default, this parameter is set to **True**.

Step 5 Click **Save**.

Configure Mobility Service Parameters

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

Procedure

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

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

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

Step 4 Configure any service parameters that you want to edit. The Mobility-related parameters are listed under the following headings. For help descriptions, click the parameter name:

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

Step 5 Click **Save**.

Configure Cisco Jabber Dual-Mode

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

Procedure

	Command or Action	Purpose
Step 1	Configure a Mobility Profile, on page 30	Configure a mobility profile to send consistent caller ID to Jabber mobile clients that are placing Dial through Office calls.
Step 2	Add a Dual-Mode Device for Cisco Jabber, on page 31	Configure a dual-mode device type for Cisco Jabber on iPhone or Android clients.

	Command or Action	Purpose
Step 3	Configure a Mobility Identity, on page 34	Add a Mobility Identity to the Jabber mobile client that points to the device phone number (that is, the iPhone number) to provide calling when Jabber roams out of WiFi range. Enable Single Number Reach destination for the Mobile Identity.
Step 4	Required: Configure Handoff Number, on page 34	Configure a handoff number for dual-mode devices that are leaving enterprise. Even when the device disconnects from the enterprise WiFi network the call can be maintained without interruption by reconnecting to a remote mobile or cellular network.

Configure Other Dual-Mode Devices

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

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

Procedure

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

Configure a Mobility Profile

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

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Mobility** > **Mobility Profile**.

Step 2 Click **Add New**.

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

Step 4 From the **Mobile Client Calling Option** drop-down list, select Dial via Office Reverse.

Note
Despite the field options, Dial via Office Forward is not available.

Step 5 Configure a **Callback Caller ID** for Dial-via-Office Reverse.

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

Step 7 Click **Save**.

Add a Dual-Mode Device for Cisco Jabber

Use the following procedure to configure a dual-mode device type for Cisco Jabber on iPhone or Android clients.

Before you begin

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

Procedure

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

Step 2 Perform one of the following:

- Click **Find** to edit an existing device.
- Click **Add New** and select either **Cisco Dual Mode for Android** or **Cisco Dual Mode for iPhone** as the phone model, to add a new device. **Click Next**.

Step 3 Configure the fields in the **Phone Configuration** window.

For detailed information about product specific configuration layout fields, see your Jabber client documentation at http://www.cisco.com/go/jabber.

Step 4 Configure the following mandatory fields:

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

Step 5 Click **Save**.

Step 6 Add a directory number:

a) In the left Association area, click **Add a New DN**.
b) Enter a new **Directory Number** and click **Save**.
c) Complete any fields that you want in the **Directory Number Configuration** window and click **Save**. For more information on the fields and their configuration options, see Online Help.
d) Click **Associate End Users** .
e) Click **Find** and select the mobility-enabled end user whom owns this DN.
f) Click **Add Selected**.
g) Click **Save**.

What to do next

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

Optionally, add Remote Destinations and Single Number Reach to your Cisco Jabber client. When someone calls the Jabber client, the remote destination rings as well. Configure a Remote Destination, on page 24.

Dual-Mode Device Configuration Fields

Table 2: Dual-Mode Device Configuration Fields

Field	Description
Softkey Template	Choose the Mobility Softkey template.
Owner User ID	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.
Mobility User ID	Choose the user ID of the person to whom this dual-mode phone is assigned.

Field	Description
Device Security Profile	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 then apply it to the phone.
Rerouting Calling Search Space	Choose a calling search space for routing calls to configured remote destinations and mobility identities that are configured for this device.
SIP Profile	Choose **Standard SIP Profile for Mobile Device**.

Add Other Dual-Mode Device

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

Before you begin

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

Procedure

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

Step 2 Click **Add New**.

Step 3 From the **Phone Model** drop-down list **Carrier-integrated Mobile Device** or **IMS-integrated Mobile Device**.

Step 4 Configure the following mandatory fields:

- Device Name
- Device Pool
- Owner User ID—The user must have mobility enabled.
- Mobility User ID—The user must have mobility enabled.

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

Step 6 Click **Save**.

Step 7 Add a directory number:

a) In the left Association area, click **Add a New DN**.
b) Enter a new **Directory Number** and click **Save**.
c) Complete any fields that you want in the **Directory Number Configuration** window and click **Save**. For more information on the fields and their configuration options, see Online Help.
d) Click **Associate End Users**.
e) Click **Find** and select the mobility-enabled end user whom owns this DN.
f) Click **Add Selected**.

g) Click **Save**.

Configure a Mobility Identity

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

Procedure

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

Step 2 Enter search criteria if needed, click **Find**, and choose the dual-mode device that you created.

Step 3 Click **Add New Mobility Identity**.

Step 4 In the **Destination** field, enter the phone number of the mobile device. For example, for a Cisco Jabber on iPhone client, this would be the phone number of the iPhone.

Step 5 Cisco Jabber only. Select the **Mobility Profile** that you configured.

Step 6 If you want to make this Mobile Identity available from an enteprise phone number:

a) Check the **Enable Single Number Reach** check box.

b) Configure a **Single Number Reach Voicemail** policy

Step 7 Configure a **Dial-via-Office Reverse Voicemail** policy.

Step 8 Configure the fields on the **Mobility Identity Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 9 Click **Save**.

Note
If you want to apply a Ring Schedule and access list to limit calls to this mobile identity to specific times and users, Configure an Access List, on page 25.

Configure Handoff Number

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

Procedure

Step 1 From Cisco Unified CM 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**.

Cisco Unified Mobility Call Flow

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

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

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

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

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

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

FMC Over SIP Trunks Without Smart Client

Unified Communications Manager allows service providers to provide base PBX-extension features such as Enterprise Dialing, SNR, Single VM, Call move, and Mid-call features through the trunk without a smart client on the mobile. Basic mobile features such as SNR, Deskphone Pickup, Send Call to Mobile, Mobile Voice Access, and Mid-call DTMF features are supported. Extension dialing is supported if it is implemented in the network and the network is integrated with Unified Communications Manager. These features can be provided by any type of trunk.

Unified Communications Manager can be configured in the Ring All Shared Lines service parameter so that the shared-line is rung when mobile DN is dialed.

Note The Reroute Remote Destination Calls to Enterprise Number feature must be enabled for Ring All Shared Lines to take effect. Reroute Remote Destination Calls to Enterprise Number is disabled by default.

IMS shared lines will ring solely based on the value of the Ring All Shared Lines parameter.

You can also migrate from the Remote Destination feature used in previous versions to this new device type.

Hunt Group Login and Logout for Carrier-Integrated Mobile Devices

When configuring the device type Carrier-Integrated Mobile, set the Owner User ID value to the mobile user identity. The mobile user identity does not appear on the configuration. Only users with mobility enabled will appear in the **Owner User ID** drop-down on the end user page and one line (DN) can be associated with an FMC device. Users should associate a mobile identity with the FMC. This can be done on the FMC device configuration page after the device has been added. For calls to be extended to the number of the mobile identity, users must enable Cisco Unified Mobility on the **Mobile Identity** window.

Carrier-integrated mobile devices can be configured to support hunt group login and logout through Enterprise Feature Access codes. Make sure that you've configured the following:

- Enterprise Feature Access must be configured in **Call Routing** > **Mobility** > **Enterprise Feature Access Number Configuration**.
- Make sure you assign values for the **Enterprise Feature Access Number for Hunt Group Login** and **Enterprise Feature Access Number for Hunt Group Logout** fields in Service Parameters.

After you configure these, the user can log in or log out of Hunt groups from Carrier-Integrated Mobile devices by dialing the configured Enterprise Feature Access Number. If the user dials the given Hunt login access code number, the Carrier-Integrated Mobile device allows them to be part of the Hunt group list. If the Hunt logout access code is dialed, then the user is moved out of the Hunt group list and calls do not reach them.

Note Users on Carrier-Integrated Mobile devices can invoke midcall features via Enterprise Feature Access codes. For details on how to configure and use Enterprise Feature Access, see Configure Enterprise Feature Access section.

Cisco Unified Mobility Interactions

Table 3: Cisco Unified Mobility Interactions

Feature	Interaction
Auto Call Pickup	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.
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 and reroutes the call through the PSTN so the caller does not need to hang up and redial. • If a rejection occurs because of resource reservation protocol (RSVP), however, AAR is not triggered for calls to remote destinations and the call stops.
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.
External Call Control	If external call control is configured, 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 Unified Communications Manager does not send a routing query for the following Cisco Unified Mobility features: • Cell pickup • Desk pickup • Session handoff

Feature	Interaction
Intelligent Session Control and Session Handoff	For direct calls to remote destinations that are anchored to the enterprise number, mobile users can use the session handoff feature to hand off the call to their deskphones. 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. **Note** Local Route Group is not supported when the AgentGreeting with BiB (Built in Bridge) is invoked.
Number of Supported Calls	Each remote destination supports a maximum of six active calls. However, the number of supported calls depends on the Unified Communications Manager configuration. For example, the Cisco Unified Mobility user receives a call while the user already has six calls for the remote destination or while the user is using DTMF to transfer or conference a call from the remote destination. The received call is sent to the enterprise voice mail when: • The number of calls with user exceeds Busy trigger configuration • CFB is configured • All shared lines are busy **Note** The calls sent to the enterprise voice mail is not based on the maximum supported calls.
SIP Trunks with Cisco Unified Border Element	Cisco Unified Mobility supports the Cisco Unified Mobility feature without midcall features over SIP trunks with Cisco Unified Border Element (CUBE).

Cisco Unified Mobility Restrictions

Table 4: Cisco Unified Mobility Interactions

Restriction	Description
Auto Answer	A remote destination call does not work when auto answer is enabled. **Note** Auto Answer is not supported with Dual-Mode phones.

Restriction	Description
Call Forwarding Unregistered	Call Forward Unregistered (CFUR) support for Cisco Jabber on iPhone and Android is as follows: • CFUR is supported if Cisco Jabber on iPhone or Android does not have either a mobile identity or remote destination configured. • CFUR is not supported, and will not work, if a Remote Destination is configured • CFUR is not supported, and will not work, if a Mobile Identity is configured with a mobile phone number and Single Number Reach is enabled. If you have a mobile identity or remote destination configured, use Call Forward Busy and Call Forward No Answer instead.
Call Queuing	Unified Communications Manager does not support call queuing with Cisco Unified Mobility.
Conferencing	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.
Dialing + Character from Mobile Phones	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.
Do Not Disturb on the Desk Phone and Direct Calls to Remote Destination	If do not disturb (DND) is enabled on a desk phone, the desk phone cannot be placed in the remote In use state and the call is not anchored in the following scenarios: • DND is enabled with the call reject option. • DND is activated by pressing the DND softkey on the desk phone. If DND is enabled with the ring off option, however, the call is anchored.

Restriction	Description
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 nor 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, Unified Communications Manager determines the registration period by using the value in the **Timer Register Expires (seconds)** field of the SIP profile that associates with the phone, not the value that the **SIP Station KeepAlive Interval** service parameter specifies. The standard SIP profile for mobile devices determines the registration period as defined by the **Time Register Expires** field in that profile.
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	**Important** From Release 15SU4 onwards, Cisco Unified Communications Manager does not support Mobile Voice Access via H.323 and SIP VoiceXML (VXML) Gateways. The Cisco Unified Mobile Voice Access Service is deprecated. Cisco Unified Mobility features are not supported for T1 CAS, FXO, FXS and BRI. SNR (Single Number Reach) is not supported with MGCP (Media Gateway Controlled Protocol).
Jabber Devices	When initially configured, Jabber devices count as registered devices. These devices increase the count of registered devices in a node, set by the **Maximum Number of Registered Devices** service parameter.
Locales	Cisco Unified Mobility supports a maximum of nine locales. If more than nine locales are installed, they appear in the Available Locales pane, but you can only save up to nine locales in the Selected Locales pane. If you attempt to configure more than nine locales for Cisco Unified Mobility, the following message appears: “Update failed. Check constraint (informix.cc_ivruserlocale_orderindex) failed.”

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.
Overlap Sending	Overlap sending patterns are not supported for the Intelligent Session Control feature.
Q Signaling	Mobility does not support Q signaling (QSIG).
QSIG Path Replacement	QSIG path replacement is not supported.
Service Parameters	Enterprise feature access service parameters apply to standard phones and smartphones; however, smartphones generally use one-touch keys to send the appropriate codes. You must configure any smartphones that will be used with Cisco Unified Mobility to use either the default codes for enterprise feature access or the codes that are specified in the smartphone documentation.
Session Handoff	The following limitations apply to the session handoff feature: • Session handoff can take place only from mobile phone to desk phone. For session handoff from desk phone to mobile phone, the current remote destination pickup method specifies that you must use send call to mobile phone. • Only audio call session handoff is supported.
Single Number Reach with Hunt Groups	If you have a hunt group configured and one or more of the directory numbers that the hunt group points toward also has Single Number Reach (SNR) enabled, the call does not extend to the SNR remote destinations unless all devices in the hunt group are logged in. For each device within the hunt group, the **Logged Into Hunt Group** check box must be checked within the **Phone Configuration** window for that device.
SIP Trunks	The Cisco Unified Mobility feature is supported only for primary rate interface (PRI) public switched telephone network (PSTN) connections. For SIP trunks, Cisco Unified Mobility is supported over IOS gateways or intercluster trunks.

Restriction	Description
SIP URI and Direct Calls to Remote Destination	The Intelligent session control feature does not support direct URI dialing. Therefore, calls that are made to a SIP URI cannot be anchored to an enterprise number.
Unified Communications Manager publisher dependent features	In a cluster environment, the publisher must be reachable in order to enable or disable Single Number Reach. Some features may not function if the publisher is not actively running. Mobile voice access is not available when the publisher node is not reachable; IVR prompts for Mobile Voice Access are stored only on the publisher.
Video Calls	Cisco Unified Mobility services do not extend to video calls. A video call that is received at the desk phone cannot be picked up on the mobile phone.

Related Topics

Ad Hoc Conferencing Service Parameters, on page 224

Cisco Unified Mobility Troubleshooting

Cannot Resume Call on Desktop Phone

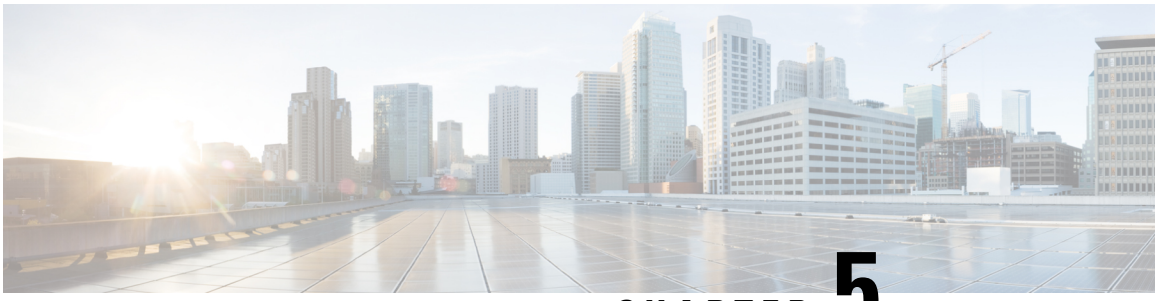
Problem When a remote destination (mobile phone) is not a smartphone and a call to this mobile phone is anchored through Cisco Unified Communications Manager, the user can hang up the mobile phone and expect to see a **Resume** softkey on the user desktop phone to resume the call. The user cannot resume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

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

- Add the following command to the gateway:

```
voice call disc-pi-off
```

- For the Cisco CallManager service, set the **Retain Media on Disconnect with PI for Active Call** service parameter to **False**.



CHAPTER 5

Device Mobility

- Device Mobility Overview, on page 43
- Device Mobility Prerequisites, on page 47
- Device Mobility Configuration Task Flow, on page 48
- Device Mobility Interactions, on page 52
- Device Mobility Restrictions, on page 53

Device Mobility Overview

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

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

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

Site-Specific Settings

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

- Date/Time Group
- Region
- Location
- Network Locale

- SRST Reference
- Connection Monitor Duration
- Physical Location
- Device Mobility Group
- Media Resource Group List

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

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

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



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

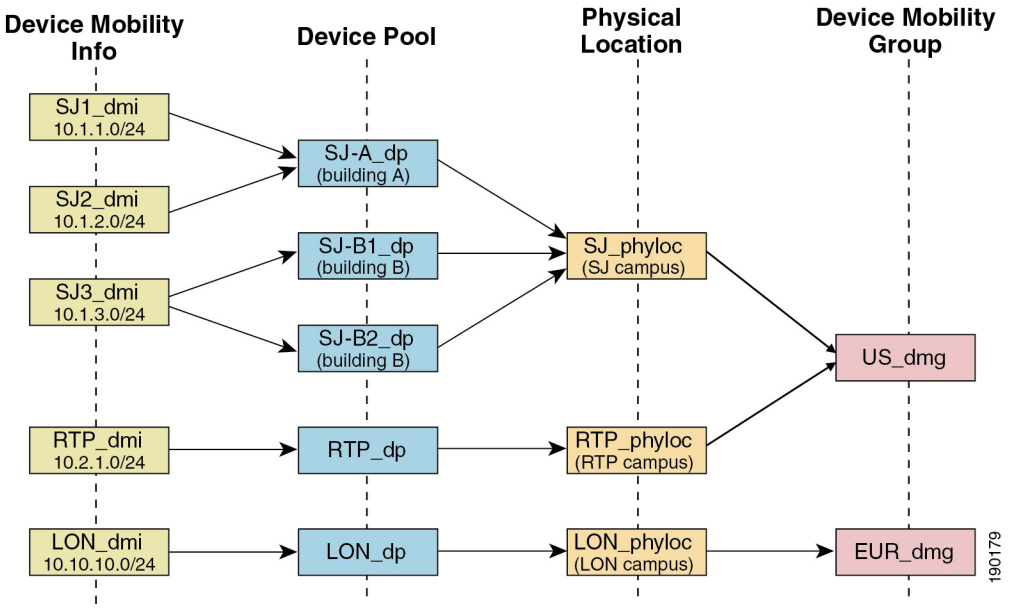
Configuration

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

- Physical Location—The physical location of the device pool. During registration, the system matches the device registration location to a subnet in the Device Mobility Info in order to assign an appropriate device pool.
- Device Pool—Location-specific device settings such as media resources, regions, and SRST references. For roaming devices, the system assigns the device pool that matches that device's physical location.
- Device Mobility Group—A logical group of sites with similar dialing patterns. For example, an enterprise with a worldwide network might set up groups that represent individual countries. The device mobility group setting determines whether the device is moved within the same geographical entity, primarily to allow users to keep their own dial plans.
- Device Mobility Info—This info contains the subnets that the system provides for roaming devices, and the device pools that the system can assign to roaming devices that register to one of those subnets.

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

Figure 1: Device Mobility-Related Configurations



Device Pool Assignment

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

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

1. A phone device record gets created for an IP phone that is provisioned to be mobile, and the phone gets assigned to a device pool. The phone registers with Unified Communications Manager, and an IP address gets assigned as part of the registration process.
2. Unified Communications Manager compares the IP address of the device to the subnets that are configured for device mobility in the Device Mobility Info Configuration window. The best match uses the largest number of bits in the IP subnet mask (longest match rule). For example, the IP address 9.9.8.2 matches the subnet 9.9.8.0/24 rather than the subnet 9.9.0.0/16.
3. If the device pool in the phone record matches the device pool in the matching subnet, the system considers the phone to be in its home location, and the phone retains the parameters of its home device pool.
4. If the device pool in the phone record does not match the device pools in the matching subnet, the system considers the phone to be roaming. The following table describes possible scenarios for device mobility and the system responses.

Table 5: Device Mobility Scenarios

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

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

Device Mobility Groups Operations Summary

You can use device mobility groups to determine when a device moves to another location within a geographic entity, so a user can use its own dial plan. For example, you can configure a device mobility group for the United States and another group for the United Kingdom. If a phone moves into a different mobility group (such as from the United States to the United Kingdom), Unified Communications Manager uses the Calling Search Space, AAR Group and AAR CSS from the phone record, and not from the roaming location.

If the device moves to another location with same mobility group (for example, Richardson, USA, to Boulder, USA), the CSS information gets taken from the roaming device pool settings. With this approach, if the user is dialing PSTN destinations, the user reaches the local gateway.

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

Table 6: Device Mobility Group Scenarios

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

Device Mobility Prerequisites

- The phone must have a dynamic IP address to use device mobility. If a phone with a static IP address roams, Unified Communications Manager uses the configuration settings from its home location.
- The Device Mobility feature requires you to set up device pools with site-specific settings. This chapter describes only the device pool settings that relate to device mobility. For more detailed information on configuring device pools, see the "*Configure Device Pools*" chapter in the System Configuration Guide for Cisco Unified Communications Manager.
- Cisco Database Layer Monitor service must be running on the same node as the Cisco CallManager service.
- Cisco TFTP service must be running on at least one node in the cluster.

- Cisco Unified Communications Manager Locale Installer (if you want to use non-English phone locales or country-specific tones).
- Any phone that runs either SCCP or SIP.

Device Mobility Configuration Task Flow

Complete these tasks to configure device mobility.

Procedure

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

Enable Device Mobility Clusterwide

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

Procedure

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

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

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

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

Step 5 Click **Save**.

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

What to do next

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

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

Enable Device Mobility for Individual Devices

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

Procedure

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

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

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

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

Step 4 Click **Save**.

Configure a Physical Location

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

Procedure

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

Step 2 Click **Add New**.

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

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

Step 5 Click **Save**.

Configure a Device Mobility Group

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

Procedure

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

Step 2 Click **Add New**.

Step 3 Enter a **Name** for the device mobility group.

Step 4 Enter a **Description** for the device mobility group.

Step 5 Click **Save**.

Configure a Device Pool for Device Mobility

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

Procedure

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

Step 2 Do either of the following:

- Click **Find** and select an existing device pool.
- Click **Add New** to create a new device pool.

Step 3 Under **Roaming Sensitive Settings**, assign the parameters that you set up in the previous device mobility tasks:

- **Physical Location**—From the drop-down list, select the physical location that you set up for this device pool. Device mobility uses this location to assign a device pool for a roaming device.
- **Device Mobility Group**—From the drop-down list, select the device mobility group that you set up for this device pool.

Step 4 Under **Device Mobility Related Information**, configure the following device mobility-related fields. For more information on the fields and their configuration options, see Online Help.

- **Device Mobility Calling Search Space**—Select the CSS to be used by a roaming device that uses this device pool.
- **AAR Calling Search Space**—Select the calling search space for the device to use when automated alternate routing (AAR) is performed.
- **AAR Group**—If AAR is configured, select the AAR Group for this device.
- **Calling Party Transformation CSS**—Select the calling party transformation CSS for roaming devices that use this device pool.

Note

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

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

Step 6 Click **Save**.

Configure Device Mobility Information

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

Procedure

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

Step 2 Click **Add New**.

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

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

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

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

Step 6 Click **Save**.

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

View Roaming Device Pool Parameters

Use the following procedure to view and verify the current device mobility settings for a device.

Procedure

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

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

Step 3 Click **View Current Device Mobility Settings** next to the **Device Mobility Mode** field.

The roaming device pool settings appear. If the device is not roaming, the home location settings appear.

Device Mobility Interactions

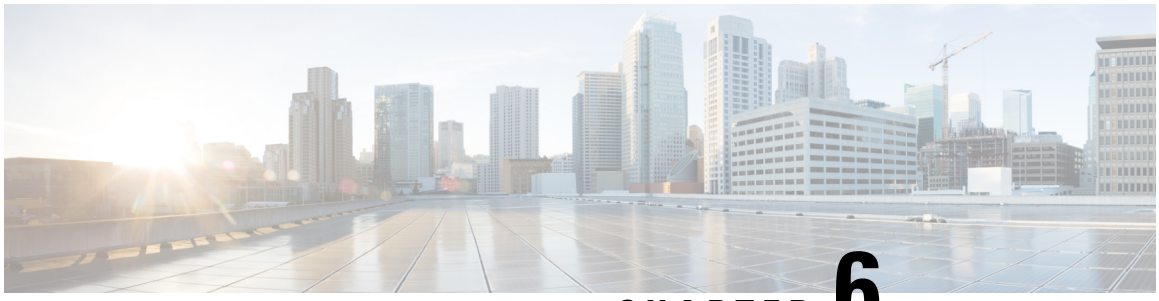
Table 7: Device Mobility Interactions

Feature	Interaction
Calling Party Normalization	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.
Roaming	When a device is roaming in the same device mobility group, Unified Communications Manager uses the device mobility CSS to reach the local gateway. If a user sets call forward all (CFA) at the phone, the CFA CSS is set to None, and the CFA CSS activation policy is set to With Activating Device/Line CSS, then the following behaviors will occur, depending on the device location: • The Device CSS and Line CSS are used as the CFA CSS when the device is in its home location. • If the device is roaming within the same device mobility group, the device mobility CSS from the roaming device pool and the line CSS are used as the CFA CSS. • If the device is roaming within a different device mobility group, the Device CSS and Line CSS are used as the CFA CSS.

Device Mobility Restrictions

Table 8: Device Mobility Restrictions

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



CHAPTER 6

Extend and Connect

- Extend and Connect Overview, on page 55
- Extend and Connect Prerequisites, on page 56
- Extend and Connect Configuration Task Flow, on page 56
- CTI Remote Device (CTIRD) Call Flows, on page 61
- Extend and Connect Interactions, on page 62
- Extend and Connect Restrictions, on page 63

Extend and Connect Overview

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

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

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

- 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 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 Installation and Configuration Guide.

Before you begin

Procedure

	Command or Action	Purpose
Step 1	Configure User Account, on page 56	Enable mobility for users so that they can use CTI remote devices. CTI devices are off-cluster phones that work with Cisco UC applications.
Step 2	Add User Permissions, on page 57	Add access control group permissions.
Step 3	Create CTI Remote Devices, on page 58	Configure off-cluster phones that users can use with Cisco UC applications.
Step 4	Add Directory Number to a Device, on page 58	Associate a directory number with the CTI remote device.
Step 5	Add Remote Destination, on page 59	Add a numerical address or directory URI that represents the other phones that the user owns.
Step 6	Verify Remote Destination, on page 60	Verify if the remote destination is successfully added for a user.
Step 7	Associate User with Device, on page 60	Associate an end user account to the CTI remote device.

Configure User Account

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

Procedure

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

Step 2 Perform either of the following:

- Click **Add New**, to configure a new user.
- Apply the filters using the **Find User Where** field and click **Find** to retrieve a list of users.

Note
You may add the new end user account through LDAP integration or local configuration.

Step 3 Locate the **Mobility Information** section.

Step 4 Check the **Enable Mobility** check box.

Step 5 Click **Save**.

Add User Permissions

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

Procedure

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

Step 2 Specify the appropriate filters in the **Find User Where** field, and then select **Find** to retrieve a list of users.

Step 3 Select the user from the list.

Step 4 Locate the **Permissions Information** section.

Step 5 Click **Add to Access Control Group**.

The **Find and List Access Control Groups** window appears.

Step 6 Click **Find**.

The Access Control Group list for Standard Users appears.

Step 7 Check the check boxes next to the following permissions:

- Standard CCM End-Users
- Standard CTI Enabled

Step 8 Click **Add Selected**.

Step 9 Click **Save**.

Create CTI Remote Devices

Use the following procedure to create a CTI remote device is a device type that represents off-cluster phones that users can use with Cisco UC applications. The device type is configured with one or more lines (directory numbers) and one or more remote destinations.

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

Procedure

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

Step 2 Click **Add New**.

Step 3 Select **CTI Remote Device** from the **Phone Type** drop-down list and then click **Next**.

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

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

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

Step 5 Edit the default value in the **Device Name** field, if appropriate.

Step 6 Enter a meaningful description in the **Description** field.

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

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

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

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

Step 9 Click **Save**.

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

Add Directory Number to a Device

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

Note

- The Calling Search Space (CSS) and partition of DN are mandatory on devices.
- The CTI Remote Device should not block its own DN. The CSS is important for the CTIRD device to reach its own DN.

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

Procedure

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

Step 2 Click **Add a new DN**.

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

Step 4 Configure all other required fields. For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save**.

Add Remote Destination

Use the following procedure to add 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.

Note Unified Communications Manager users can add remote destinations through the Cisco Jabber interface. For more information, see the Cisco Jabber for Windows Installation and 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 CM Administration, choose **Device** > **Phone**.

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

Step 3 Select the CTI remote device from the list.

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

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

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

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 CM Administration, choose **Device** > **Phone**.

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

Step 3 Select the CTI remote device from the list.

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

Step 5 Click **Apply Config**.

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

Associate User with Device

Procedure

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

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

Step 3 Select the user from the list.

Step 4 Locate the **Device Information** section.

Step 5 Click **Device Association**.

Step 6 Find and select the CTI remote device.

Step 7 To complete the association, click **Save Selected/Changes**.

Step 8 From **Related Links** drop-down list, choose **Back to User**, and then click **Go**.
The **End User Configuration** window appears, and the associated device that you chose appears in the **Controlled Devices** pane.

CTI Remote Device (CTIRD) Call Flows

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

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

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

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

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

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

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

Extend and Connect Interactions

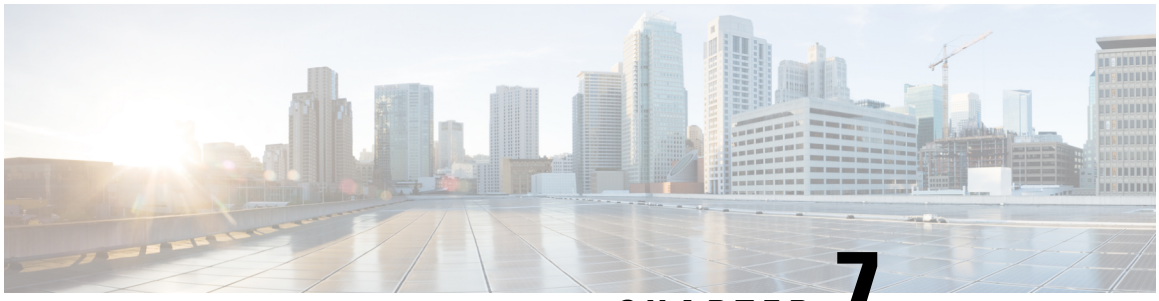
Table 9: Extend and Connect Interactions

Feature	Interaction
Directory URI Dialing	Configure a Directory URI as the DN, remote destination, or both for the CTI remote device.
Unified Mobility	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.
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 IP Phone is available to answer hunt calls (logged-in/HLog). • Cisco Jabber is running in Extend and Connect mode.
CallerID Information	• 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 10: Extend and Connect Restrictions

Restriction	Description
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** Two or more users cannot use the same remote destination numbers.
Remote destination validation	• Remote destination numbers are validated using the CTI remote device reroute calling search space. • Remote destinations that are configured using the Cisco Unified Communications Manager Administration interface and AXL interface are not validated.
Consult transfer limitation	When a consult transfer is initiated from a CTI Remote Device to an internal IP phone or another extend and connect enabled device, no ringback is heard on the remote destination associated to the CTI Remote device which is initiating the transfer.
Call Forward Unregistered	Extend and Connect does not support Call Forward Unregistered Internal or Call Forward Unregistered External.
Route Next Hop By Calling Party Number	Extend and Connect does not support Translation Patterns when the "Route Next Hop By Calling Party Number" option is enabled.



CHAPTER 7

Remote Worker Emergency Calling

- Remote Worker Emergency Calling Overview, on page 65
- Remote Worker Emergency Calling Prerequisites, on page 65
- Remote Worker Emergency Calling Configuration Task Flow, on page 66

Remote Worker Emergency Calling Overview

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

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

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

Remote Worker Emergency Calling Prerequisites

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

Remote Worker Emergency Calling Configuration Task Flow

Before you begin

Procedure

	Command or Action	Purpose
Step 1	Configure User As a Remote Worker, on page 66	Associate the off-premises device with the owner of the device.
Step 2	Specify Alternate Routing for Emergency Calling, on page 67	These parameters specify the calling search space and destination number that are used to restrict the routing of any call that is 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.
Step 3	Configure the Application Server, on page 67	Direct end users to the application server where they enter the location of the device.
Step 4	Configure E911 Messages, on page 67	Configure the E911 messages that appear on an off-premises end-user phone.

Configure User As a Remote Worker

Before you begin

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

Procedure

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

Step 2 Enter the appropriate search criteria to find the phone and click **Find**.
A list of phones that match the search criteria is displayed.

Step 3 Choose the phone for which you want to configure Remote Worker Emergency Calling.
The **Phone Configuration** window is displayed.

Step 4 From the **Device Information** section, select the appropriate user ID from the **Owner User ID** drop-down list and check the **Require off-premise location** check box.

Step 5 Click **Save**.

Specify Alternate Routing for Emergency Calling

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

Procedure

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

Step 2 From the **Server** drop-down list, choose a server.

Step 3 From the **Service** drop-down list, choose **Cisco CallManager**.
The **Service Parameter Configuration** window appeares.

Step 4 In the **Clusterwide Parameters (Emergency Calling for Required Off-premise Location)** section, specify **Alternate Destination for Emergency Call**.

Step 5 Specify **Alternate Calling Search Space for Emergency Call**.

Step 6 Click **Save**.

Configure the Application Server

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Application Server**.

Step 2 Click **Add New**.
The **Application Server** window appears.

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**.

Configure E911 Messages

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

Procedure

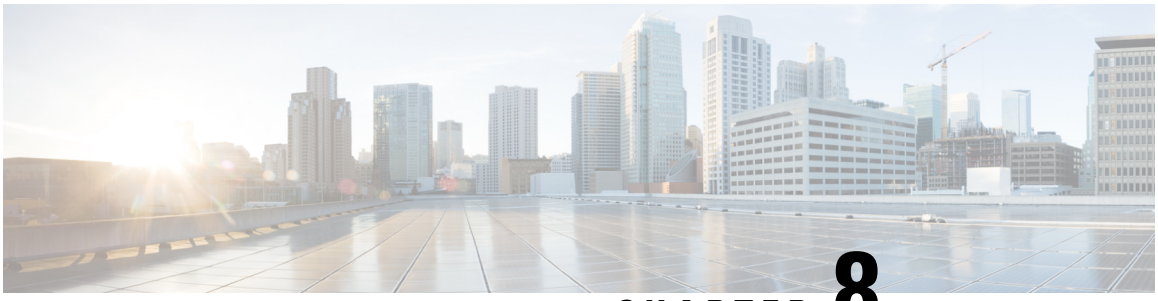
Step 1 From Cisco Unified CM Administration, choose **System** > **E911 Messages**.

Step 2 Select the required language link of the E911 messages.

The **E911 Messages Configuration** page displays the Agreement, Disclaimer, and Error messages.

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

Step 4 Click **Save**.



CHAPTER 8

Configure Mobile and Remote Access

- Mobile and Remote Access Overview, on page 69
- Mobile and Remote Access Prerequisites, on page 71
- Mobile and Remote Access Configuration Task Flow, on page 72
- MRA Failover with Lightweight Keepalives, on page 78

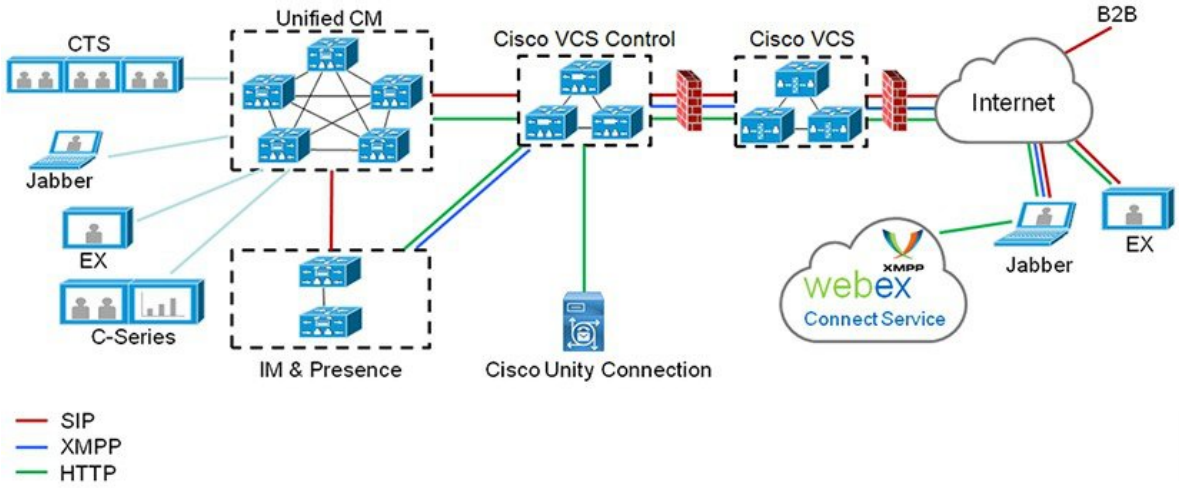
Mobile and Remote Access Overview

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

The overall solution provides:

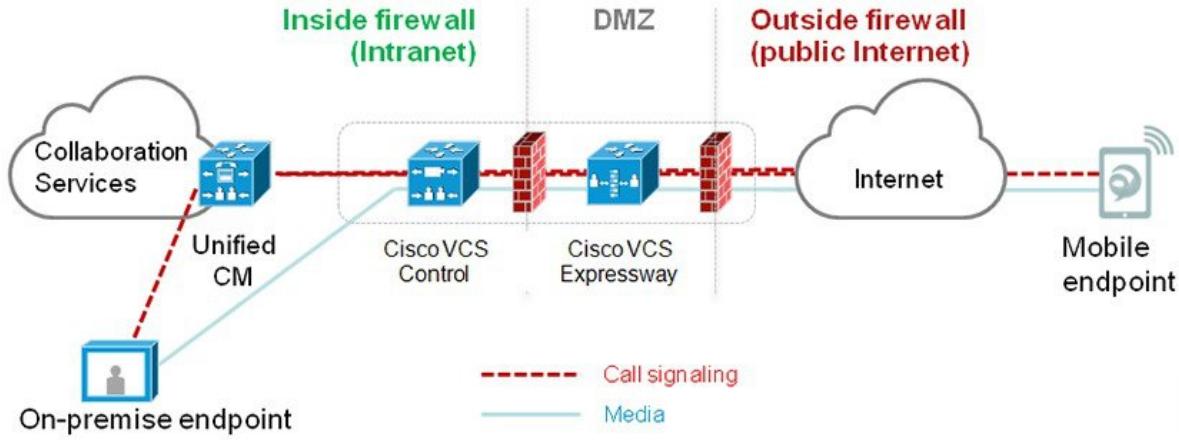
- Off-premises access: a consistent experience outside the network for Jabber and EX/MX/SX Series clients
- Security: secure business-to-business communications
- Cloud services: enterprise grade flexibility and scalable solutions providing rich Webex integration and Service Provider offerings
- Gateway and interoperability services: media and signaling normalization, and support for non-standard endpoints

Figure 2: Unified Communications: Mobile and Remote Access



Third-party SIP or H.323 devices can register to the Expressway-C and, if necessary, interoperate with Unified CM-registered devices over a SIP trunk.

Figure 3: Typical Call Flow: Signaling and Media Paths



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

Configuring Mobile and Remote Access

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

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

Mobile and Remote Access Prerequisites

Cisco Unified Communications Manager Requirements

The following requirements apply:

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

DNS Requirements

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

`_cisco-uds._tcp<domain>`

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

Cisco Expressway Requirements

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

The minimum Expressway release for Mobile and Remote Access Access Policy support with Cisco Jabber is X8.10.

Certificate Prerequisites

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

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

Note If you use different CAs, you must install each CA's root certificate chain on Unified Communications Manager, IM and Presence Service Service, and Expressway-C.

Note You can also use self-signed certificates for both Unified Communications Manager and the IM and Presence Service Service. In this case, you must upload onto Expressway-C the tomcat and Cisco CallManager certificates for Unified Communications Manager and a tomcat certificate for the IM and Presence Service Service.

Mobile and Remote Access Configuration Task Flow

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

Procedure

	Command or Action	Purpose
Step 1	Activate Cisco AXL Web Service, on page 73	Make sure that the Cisco AXL Web Service is activated on the publisher node.
Step 2	Configure Maximum Session BitRate for Video, on page 73	**Optional**. Configure Region-specific settings for your Mobile and Remote Access endpoints. For example, if you expect Mobile and Remote Access endpoints to use video, you may want to increase the **Maximum Session Bit Rate for Video Calls** setting as the default setting of 384 kbps may be too low for some video endpoints.
Step 3	Configure a Device Pool for Mobile and Remote Access, on page 74	Assign your Date/Time Group and Region configuration to the device pool that your Mobile and Remote Access endpoints use.
Step 4	Configure ICE, on page 74	**Optional**. ICE is an optional deployment that uses STUN and TURN services to analyze the available media paths for an Mobile and Remote Access call and then to select the best path. ICE adds potentially to the call setup time, but increases the reliability of Mobile and Remote Access calls.
Step 5	Configure Phone Security Profile for Mobile and Remote Access, on page 75	Use this procedure to set up a phone security profile to be used by Mobile and Remote Access endpoints.
Step 6	Configure Mobile and Remote Access Access Policy for Cisco Jabber Users, on page 76	Cisco Jabber only. Set up an Mobile and Remote Access Access Policy for Cisco Jabber users. Cisco Jabber users must be enabled with Mobile and Remote Access access within their user profiles in order to use the Mobile and Remote Access feature.

	Command or Action	Purpose
Step 7	Configure Users for Mobile and Remote Access, on page 77	For Cisco Jabber users, the User Policy that you set up must be applied to their End User Configurations.
Step 8	Configure Endpoints for Mobile and Remote Access, on page 77	Configure and provision endpoints that use the Mobile and Remote Access feature.
Step 9	Configure Cisco Expressway for Mobile and Remote Access, on page 78	Configure Cisco Expressway for Mobile and Remote Access.

Activate Cisco AXL Web Service

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

Procedure

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

Step 2 From the **Server** drop-down list, select the publisher node and click **Go**.

Step 3 Under **Database and Admin Services**, confirm that the **Cisco AXL Web Service** is **Activated**.

Step 4 If the service is not activated, check the corresponding check box and click **Save** to activate the service.

Configure Maximum Session BitRate for Video

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Region Information** > **Region**.

Step 2 Perform any one of the following:

- Click **Find** and select the region to edit the bit rates within an existing region.
- Click **Add New** to create a new region.

Step 3 In the **Modify Relationship to other Regions** area, configure a new setting for the **Maximum Session Bit Rate for Video Calls**. For example, 6000 kbps.

Step 4 Configure any other fields in the **Region Configuration** window. For more information on the fields and their configuration options, see Online Help..

Step 5 Click **Save**.

Configure a Device Pool for Mobile and Remote Access

When you created a new region, assign your region to the device pool that your Mobile and Remote Access endpoints use.

Procedure

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

Step 2 Do either of the following:

- Click **Find** and select the existing device pool to edit.
- Click **Add New** to create a new device pool.

Step 3 Enter a **Device Pool Name**.

Step 4 Select a redundant **Cisco Unified Communications Manager Group**.

Step 5 Assign the **Date/Time Group** that you set up. This group includes the Phone NTP references that you set up for Mobile and Remote Access endpoints.

Step 6 From the **Region** drop-down list, select the region that you configured for Mobile and Remote Access.

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

Step 8 Click **Save**.

Configure ICE

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

Before you begin

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

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

Procedure

Step 1 From Cisco Unified CM Administration:

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

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

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

Step 4 Set the **Default Candidate Type**:

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

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

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

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

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

Step 9 Click **Save**.

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

Configure Phone Security Profile for Mobile and Remote Access

Use this procedure to set up a phone security profile to be used by Mobile and Remote Access endpoints.

Procedure

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

Step 2 Click **Add New**.

Step 3 From the **Phone Security Profile Type** drop-down list, select your device type. For example, you could select **Cisco Unified Client Service Framework** for a Jabber application.

Step 4 Click **Next**.

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

Step 6 From the **Device Security Mode** drop-down list, select **Encrypted**.

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

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

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

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

Step 10 Click **Save**.

Note
You must apply this profile to the Phone Configuration for each of your Mobile and Remote Access endpoints.

Configure Mobile and Remote Access Access Policy for Cisco Jabber Users

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

Note The Mobile and Remote Access Policy is not required for non-Jabber users.

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

Procedure

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

Step 2 Click **Add New**.

Step 3 Enter a **Name** and **Description** for the user profile.

Step 4 Assign a **Universal Device Template** to apply to users' **Desk Phones**, **Mobile and Desktop Devices**, and **Remote Destination/Device Profiles**.

Step 5 Assign a **Universal Line Template** to apply to the phone lines for users in this user profile.

Step 6 If you want the users in this user profile to be able to use the self-provisioning feature to provision their own phones, do the following:

a) Check the **Allow End User to Provision their own phones** check box.

b) In the **Limit Provisioning once End User has this many phones** field, enter a maximum number of phones the user is allowed to provision. The maximum is 20.

c) Check the **Allow Provisioning of a phone already assigned to a different End User** check box to determine whether the user who is associated with this profile has the permission to migrate or reassign a device that is already owned by another user. By default, this check box is unchecked.

Step 7 If you want Cisco Jabber users who are associated with this user profile, to be able to use the Mobile and Remote Access feature, check the **Enable Mobile and Remote Access** check box.

Note

- By default, this check box is selected. When you uncheck this check box, the **Client Policies** section is disabled, and No Service client policy option is selected by default.
- This setting is mandatory only for Cisco Jabber users whom are using OAuth Refresh Logins. Non-Jabber users do not need this setting to be able to use Mobile and Remote Access. Mobile and Remote Access feature is applicable only for the Jabber Mobile and Remote Access users and not to any other endpoints or clients.

Step 8 Assign the Jabber policies for this user profile. From the **Desktop Client Policy**, and **Mobile Client Policy** drop-down list, choose one of the following options:

- No Service—This policy disables access to all Cisco Jabber services.
- IM & Presence only—This policy enables only instant messaging and presence capabilities.
- IM & Presence, Voice and Video calls—This policy enables instant messaging, presence, voicemail, and conferencing capabilities for all users with audio or video devices. This is the default option.

Note
Jabber desktop client includes Cisco Jabber for Windows users and Cisco Jabber for Mac users. Jabber mobile client includes Cisco Jabber for iPad and iPhone users and Cisco Jabber for Android users.

Step 9 If you want the users in this user profile to set the maximum login time for Extension Mobility or Extension Mobility Cross Cluster through Cisco Unified Communications Self Care Portal, check the **Allow End User to set their Extension Mobility maximum login time** check box.

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

Step 10 Click **Save**.

Configure Users for Mobile and Remote Access

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

Configure Endpoints for Mobile and Remote Access

Provision and configure endpoints for Mobile and Remote Access:

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

Configure Cisco Expressway for Mobile and Remote Access

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

MRA Failover with Lightweight Keepalives

Important This section is applicable from Release 14 onwards.

The MRA High-Availability for endpoint registration allows Cisco Webex and Cisco Jabber to detect any failure of network elements like Cisco Expressway-E, Cisco Expressway-C, and Cisco Unified Communications Manager Administration in the Registration path and take corrective action to reregister to the Unified CM through the next available path.

The endpoints send a lightweight STUN keepalive message to check connectivity in the registration path. When the Unified Communications Manager receives the lightweight STUN keepalive message, it validates the Cisco Expressway-C IP and responds to the message. The Unified CM discards the STUN keepalive message if it is received from any other IP.

If a node in the registration path fails, endpoints will learn the failure through the lightweight STUN keepalive response that they receive and selects a different route path for future messages. This service helps the user to have smooth and continuous incoming and outgoing calls irrespective of outages or other maintenance modes.

When Cisco Webex or Cisco Jabber registers to the Unified Communications Manager as a MRA device, the system displays the Expressway-C's IP in the Unified CM (**Device** > **Phone** > **IPv4 Address** column).

Note The Cisco IP Phones do not support registration failover.

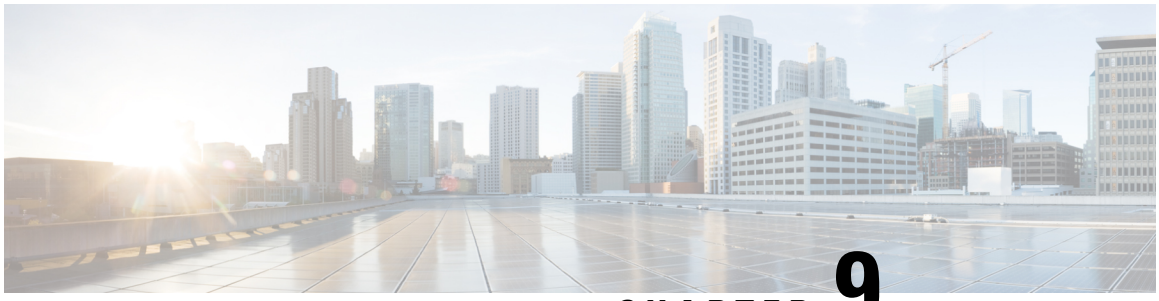
For more information, see Mobile and Remote Access Through Cisco Expressway Deployment Guide.



PART III

Remote Network Access

- Wireless LAN, on page 81
- VPN Client , on page 85



CHAPTER 9

Wireless LAN

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

Wireless LAN Overview

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

Wireless LAN Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that wireless LAN profiles.
Step 2	Configure a Network Access Profile, on page 82	**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.
Step 3	Configure a Wireless LAN Profile, on page 82	Configure a wireless LAN profile with common WiFi settings to apply to devices or device pools in the enterprise.
Step 4	Configure a Wireless LAN Profile Group, on page 82	Group wireless LAN profiles together.
Step 5	To Link a Wireless LAN Profile Group to a Device or Device Pool, on page 83, perform one of the following subtasks:	After you complete the device link, TFTP adds the wireless LAN profile group to the existing device configuration file, which the device (or

	Command or Action	Purpose
	• Link a Wireless LAN Profile Group to a Device, on page 83 • Link a Wireless LAN Profile Group to a Device Pool, on page 83	devices that are tied to a device pool) proceeds to download.

Configure a Network Access Profile

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Network Access Profile**

Step 2 Click **Add New**.

Step 3 Configure the fields in the **Network Access Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure a Wireless LAN Profile

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Wireless LAN Profile**

Step 2 Click **Add New**.

Step 3 Configure the fields in the **Wireless LAN Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure a Wireless LAN Profile Group

Group your wireless LAN profiles.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Wireless LAN Profile Group**.

Step 2 Click **Add New**.

Step 3 Configure the fields in the **Wireless LAN Profile Group Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

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

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

Procedure

	Command or Action	Purpose
Step 1	Link a Wireless LAN Profile Group to a Device, on page 83	
Step 2	Link a Wireless LAN Profile Group to a Device Pool, on page 83	

Link a Wireless LAN Profile Group to a Device

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to enter search criteria and choose an existing device from the resulting list.
- Click **Add New**, and choose the device type from the **Phone Type** drop-down list.

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

Step 4 Click **Save**.

Link a Wireless LAN Profile Group to a Device Pool

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

Procedure

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

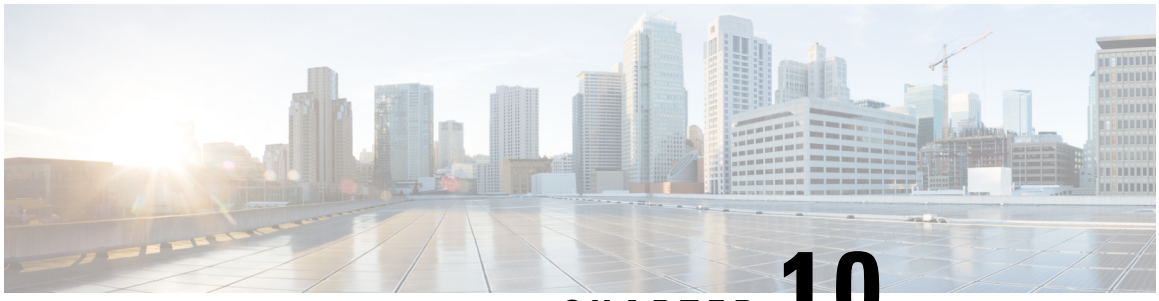
Step 2 Perform one of the following tasks:

- Click **Find** to enter search criteria and choose an existing device pool from the resulting list.

- Click **Add New**.

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

Step 4 Click **Save**.



CHAPTER 10

VPN Client

- VPN Client Overview, on page 85
- VPN Client Prerequisites, on page 85
- VPN Client Configuration Task Flow, on page 85

VPN Client Overview

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

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

VPN Client Prerequisites

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

VPN Client Configuration Task Flow

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.

Procedure

	Command or Action	Purpose
Step 1	Complete Cisco IOS Prerequisites, on page 87	Complete Cisco IOS prerequisites. Perform this action if you want to configure Cisco IOS VPN.
Step 2	Configure Cisco IOS SSL VPN to Support IP Phones , on page 87	Configure Cisco IOS for VPN client on an IP Phone. Perform this action if you want to configure Cisco IOS VPN.
Step 3	Complete ASA Prerequisites for AnyConnect, on page 89	Complete ASA prerequisites for AnyConnect. Perform this action if you want to configure ASA VPN.
Step 4	Configure ASA for VPN Client on IP Phone, on page 89	Configure ASA for VPN client on an IP Phone. Perform this action if you want to configure ASA VPN.
Step 5	Configure the VPN concentrators for each VPN Gateway.	To avoid long delays when the user upgrades the firmware or configuration information on a remote phone, set up the VPN concentrator close in the network to the TFTP or Unified Communications Manager server. If this is not feasible in your network, you can set up an alternate TFTP or load server that is next to the VPN concentrator.
Step 6	Upload VPN Concentrator Certificates, on page 92	Upload the VPN concentrator certificates.
Step 7	Configure VPN Gateway, on page 92	Configure the VPN gateways.
Step 8	Configure VPN Group, on page 93	After you create a VPN group, you can add one of the VPN gateways that you just configured to it.
Step 9	Perform one of the following: • Configure VPN Profile, on page 94 • Configure VPN Feature Parameters, on page 96	You must configure a VPN profile only if you have multiple VPN groups. The VPN Profile fields take precedence over the VPN Feature Configuration fields.
Step 10	Add VPN Details to Common Phone Profile, on page 97	Add the VPN Group and VPN Profile to a Common Phone Profile.
Step 11	Upgrade the firmware for Cisco Unified IP Phone to a version that supports VPN.	To run the Cisco VPN client, a supported Cisco Unified IP Phone must be running firmware release 9.0(2) or higher. For more information about upgrading the firmware, see *Cisco Unified IP Phone Administration Guide* for Unified Communications Manager for yourCisco Unified IP Phone model.

	Command or Action	Purpose
Step 12	Using a supported Cisco Unified IP Phone, establish the VPN connection.	Connect your Cisco Unified IP Phone to a VPN.

Complete Cisco IOS Prerequisites

Use this procedure to complete Cisco IOS Prerequisites.

Procedure

Step 1 Install Cisco IOS Software version 15.1(2)T or later.

Feature Set/License: Universal (Data & Security & UC) for IOS ISR-G2 and ISR-G3

Feature Set/License: Advanced Security for IOS ISR

Step 2 Activate the SSL VPN License.

Configure Cisco IOS SSL VPN to Support IP Phones

Use this procedure to complete Cisco IOS SSL VPN to Support IP Phones.

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 Unified Communications Manager.

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

Note

This location changes based on the Unified Communications Manager version.

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

c) Create trustpoint on the Cisco IOS software.

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

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

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

- Generate a self-signed certificate.

```
Router> enable
Router# configure terminal
Router(config)# crypto key generate rsa general-keys label <name>
<exportable -optional>Router(config)# crypto pki trustpoint <name>
Router(ca-trustpoint)# enrollment selfsigned
Router(ca-trustpoint)# rsakeypair <name> 2048 2048
Router(ca-trustpoint)#authorization username subjectname commonname
Router(ca-trustpoint)# crypto pki enroll <name>
Router(ca-trustpoint)# end
```

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

  Example:

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

- Register the generated certificate with Unified Communications Manager.

  Example:

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

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

Step 4 Install AnyConnect on Cisco IOS.

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

Example:

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

Step 5 Configure the VPN feature.

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 k1kLGQIoxyCO4ti9 encrypted
```

Complete ASA Prerequisites for AnyConnect

Use this procedure to complete ASA Prerequisites for AnyConnect.

Procedure

Step 1 Install ASA software (version 8.0.4 or later) and a compatible ASDM.

Step 2 Install a compatible AnyConnect package.

Step 3 Activate License.

a) Check features of the current license using the following command:

show activation-key detail

b) If necessary, obtain a new license with additional SSL VPN sessions and enable the Linksys phone.

Step 4 Make sure that you configure a tunnel-group with a non-default URL as follows:

```
tunnel-group phonevpn type remote-access
tunnel-group phonevpn general-attribute
  address-pool 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 Unified Communications Manager.
- It is preferred to have the certificate CN or subject alternate name match the FQDN or IP address in the group-url.
- If the ASA certificate CN or SAN does not match with the FQDN or IP address, uncheck the host ID check box in the Unified Communications Manager.

Configure ASA for VPN Client on IP Phone

Use this procedure to configure ASA for VPN Client on IP Phone.

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

Procedure

Step 1 Local configuration

a) Configure network interface.

Example:

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

b) Configure static routes and default routes.

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

Example:

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

c) Configure the DNS.

Example:

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

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

Import the following certificates from the Unified Communications Manager.

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

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

a) From the Cisco Unified OS Administration, choose **Security** > **Certificate Management**.
b) Locate the certificates Cisco_Manufacturing_CA and CAPF. Download the `.pem` file and save asa `.txt` file.
c) Create trustpoint on the ASA.

Example:

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

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

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

- Generate a self-signed certificate.

  Example:

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

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

  Example:

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

- Register the generated certificate with Unified Communications Manager.

  Example:

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

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

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

Note
To use the phone with both certificate and password authentication, create a user with the phone MAC address. Username matching is case sensitive. For example:

```
ciscoasa(config)# username CP-7975G-SEP001AE2BC16CB password k1kLGQIoxyCO4ti9 encrypted
ciscoasa(config)# username CP-7975G-SEP001AE2BC16CB attributes
ciscoasa(config-username)# vpn-group-policy GroupPhoneWebvpn
ciscoasa(config-username)#service-type remote-access
```

ASA Certificate Configuration

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

Upload VPN Concentrator Certificates

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

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

If a Locally Significant Certificate (LSC) is installed on the Cisco Unified IP Phone, it will send its LSC by default.

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

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

Procedure

Step 1 From Cisco Unified OS Administration, choose **Security** > **Certificate Management**.

Step 2 Click **Upload Certificate**.

Step 3 From the **Certificate Purpose** drop-down list, choose **Phone-VPN-trust**.

Step 4 Click **Browse** to choose the file that you want to upload.

Step 5 Click **Upload File**.

Step 6 Choose another file to upload or click **Close**.

For more information, see *Certificate Management* chapter.

Configure VPN Gateway

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 92.

Use this procedure to configure the VPN Gateway.

Procedure

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

Step 2 Perform one of the following tasks:

a) Click **Add New** to configure new profile.
b) Click the **Copy** next to the VPN gateway that you want to copy.
c) Locate the appropriate VPN gateway and modify the settings to update an existing profile.

Step 3 Configure the fields in the **VPN Gateway Configuration** window. For more information, see VPN Gateway Fields for VPN Client, on page 93.

Step 4 Click **Save**.

VPN Gateway Fields for VPN Client

The table describes the VPN Gateway fields for VPN Client.

Table 11: VPN Gateway Fields for VPN Client

Field	Description
VPN Gateway Name	Enter the name of the VPN gateway.
VPN Gateway Description	Enter a description of the VPN gateway.
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 fails 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

Use this procedure to configure VPN Group.

Procedure

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

Step 2 Perform one of the following tasks:

a) Click **Add New** to configure new profile.
b) Click **Copy** next to the VPN group that you want to copy an existing VPN group.
c) Locate the appropriate VPN group and modify the settings to update an existing profile.

Step 3 Configure the fields in the **VPN Group Configuration** window. For more information, see VPN Gateway Fields for VPN Client, on page 93 for the field description details.

Step 4 Click **Save**.

VPN Group Fields for VPN Client

The table describes the VPN Group Fields for VPN Client.

Table 12: VPN Group Fields for VPN Client

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

Configure VPN Profile

Use this procedure to configure the VPN Profile.

Procedure

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

Step 2 Perform one of the following tasks:

a) Click **Add New** to configure new profle.
b) Click **Copy** next to the VPN profile that you want to copy an existing profile.
c) To update an existing profile, specify the appropriate filters in the **Find VPN Profile Where**, click **Find**, and modify the settings.

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

Step 4 Click **Save**.

VPN Profile Fields for VPN Client

The table describes the VPN profile field details.

Table 13: VPN Profile Field Details

Field	Definition
Name	Enter a name for the VPN profile.
Description	Enter a description for the VPN profile.
Enable Auto Network Detect	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.
MTU	Enter the size, in bytes, for the Maximum Transmission Unit (MTU). Default: 1290 bytes.
Fail to Connect	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
Enable Host ID Check	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
Client Authentication Method	From the drop-down list, choose the client authentication method: • User and password • Password only • Certificate (LSC or MIC)
Enable Password Persistence	When you check this check box, a user password gets saved in the phone until either a failed log in attempt occurs, a user manually clears the password, or the phone resets or loses power.

Configure VPN Feature Parameters

Procedure

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

Step 2 Configure the fields in the **VPN Feature Configuration** window. For more information, see VPN Feature Parameters, on page 96.

Step 3 Click **Save**.

Perform the following tasks:

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

VPN Feature Parameters

The table describes the VPN feature parameters.

Table 14: VPN Feature Parameters

Field	Default
Enable Auto Network Detect	When True, the VPN client can only run when it detects that it is out of the corporate network. Default: False
MTU	This field specifies the maximum transmission unit: Default: 1290 bytes Minimum: 256 bytes Maximum: 1406 bytes
Keep Alive	This field specifies the rate at which the system sends the keep alive message. **Note** If it is non zero and less than the value specified in Unified Communications Manager, the keep alive setting in the VPN concentrator overwrites this setting. Default: 60 seconds Minimum: 0 Maximum: 120 seconds

Field	Default
Fail to Connect	This field specifies the amount of time to wait for login or connect operations to complete while the system creates the VPN tunnel. Default: 30 seconds Minimum: 0 Maximum: 600 seconds
Client Authentication Method	From the drop-down list, choose the client authentication method: • User and password • Password only • Certificate (LSC or MIC) Default: User And Password
Enable Password Persistence	When True, a user password gets saved in the phone, if Reset button or "**#**" is used for reset. The password does not get saved and the phone prompts for credentials if the phone loses power or you initiate a factory reset. Default: False
Enable Host ID Check	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

Use this procedure to add VPN details to common phone profile.

Procedure

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

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

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

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

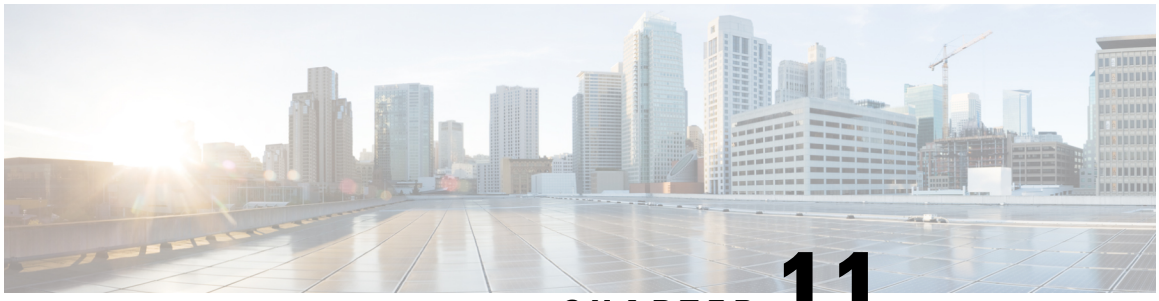
Step 5 Click **OK** in apply configuration window.



PART IV

Licensing

- Licensing, on page 101



CHAPTER 11

Licensing

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Licensing

Cisco Unified Communications Manager licensing is part of the overall commercial offer of Cisco Unified Communications Licensing.

		User Connect Licensing (Essential)	User Connect Licensing (Basic)	User Connect Licensing (Enhanced/ Enhanced Plus)	Unified Workspace Licensing
Cisco Unified CM Features	**Mobile Connect (SNR)**	Not Available	Included	Included	Included

		User Connect Licensing (Essential)	User Connect Licensing (Basic)	User Connect Licensing (Enhanced/ Enhanced Plus)	Unified Workspace Licensing
Device Support	**Number of Devices**	1	1	1/2	10
	Device Type Support	Analog/Voice (for details, see the User and Device table)	Voice (for details, see the User and Device table)	Voice (for details, see the User and Device table)	Voice (for details, see theUser and Device table)
	Number of User Profiles	1	1	1	1
Clients	**Jabber Mobile**	Not Available	Not Available	Included	Included
	Jabber Desktop	Not Available	Not Available	Included	Included
	Jabber IM/Presence	Included	Included	Included	Included
Application	**Webex Meetings**	Add-on	Add-on	Add-on	Included
	Webex Social	Add-on	Add-on	Add-on	Included
	Unity Connection	Add-on	Add-on	Add-on	Included
	Cisco Unified CM	Included	Included	Included	Included

Licensing for the Cisco Unified Communications Manager is determined by the total number of users, user features, and devices configured. Cisco Unified Communications Manager calculates its license usage based upon the total number of users (with user features and associated devices) and devices configured on the system. Cisco Unified Communications Manager reports the total license usage (per publisher) to the Cisco Smart Software Manager and gets back the license compliance or non-compliance status.

Unified Communications Manager Licensing

Cisco Unified Workspace Licensing (UWL) allows organizations to access a wide range of Cisco Collaboration applications and services in a cost-effective, simple package. It includes soft clients, application server software, and licensing on a per-user basis.

Cisco User Connect Licensing (UCL) is a user-based license for individual Cisco Unified Communications products. It includes a soft client, application server software licensing, and basic unified communications applications. Depending on your needs and device of choice, UCL is available in Essential, Basic, Enhanced, or Enhanced Plus.

The following are the license types for the Unified Communications Manager:

UC Manager Essential	Essential User Connect License - supports one device providing basic voice or analog device (phone or fax). (For example: analog phone, ATA 186, ATA 187, Cisco 3905, Cisco 6901)
UC Manager Basic	Basic User Connect License - supports one device, including all Essential devices, plus basic (voice and video) call control features. (For example: Cisco 6911, Cisco 6921, Cisco 9811)
UC Manager Enhanced	Enhanced User Connect License - supports one device, including all Basic devices, plus advanced (voice and video) call control features including desktop, mobile clients. (For example: Cisco 3911, Cisco 3951, Cisco 6941, Cisco 6945, Cisco 6961, Cisco 79xx, Cisco 89xx, Cisco 9841, 9851, 9861, and 9871, Cisco 99xx, Cisco E20, Cisco TelePresence EX60, Cisco TelePresence EX90, third party SIP **Note** CTI route points and CTI ports are covered by calling KW licenses. If your Flex Agreement does not include calling KW licenses, you must obtain separate Enhanced licenses for CTI route points and CTI ports.
UC Manager Enhanced Plus	Enhanced Plus User Connect License - supports up to two devices, and including all Enhanced devices.
UC Manager CUWL	Supports advanced (voice and video) call control features including desktop and mobile, professional collaboration workspace application features with a maximum of ten devices per user.
UC Manager TelePresence Room	TelePresence Room license - supports room based immersive and multipurpose Cisco TelePresence System endpoints and Spark Room. (For example: Cisco TelePresence System Series 3200, 3000, 1300, Cisco TelePresence MX Series, Cisco TelePresence TX Series, Cisco TelePresence System Profile Series, Cisco Expressway)

Commercial Licensing

Cisco Unified Communications Manager software and user licenses are available through Cisco Collaboration Flex Plan 3.0. Ordering options include:

- A-Flex-3
- A-FLEX-3-EDU (EOL, reaches LDOS June 30, 2026)
- A-Flex-3-FEDRAMP
- E3-COLLABORATION

Flex Plan 3.0 offers both Enterprise Agreement (EA) and Named User (NU) buying models.

Customers with older perpetual licenses (such as FLEX Subscription v10.x) should transition to On-Premises or Partner Hosted Smart Licensing under Cisco Collaboration Flex Plan 3.0. For comprehensive details on license types and ordering, refer to the Flex Plan 3.0 ordering guide.



Note Legacy license types, such as Enhanced and Enhanced Plus, are no longer valid under Flex Plan 3.0.

Cisco has announced the End-of-Sale (EoS) for perpetual licensing models for Unified Communications Manager. You can continue to use on-premises calling while migrating to Flex Plan 3.0. The following are the Flex Plan 3.0 license types for Unified Communications Manager:

Flex 3.0	License Type	Consumed by
NU Access	Basic	Device (low-end phones) User and Device (low-end phones) Cisco IP Phones 7811, 7821, 6901, Cisco Analog Telephony Adapters, Cisco Unified SIP Phone 3905, and Fax.
NU Enhanced	Enhanced	Device User and Device Cisco IP Phones 7841 and 7861, Cisco IP Phones 7945 and 7965, Cisco IP Phones 8811, 8821, 8841, 8851, 8861, 8875, Cisco Unified IP Phones 9900 Series, Jabber clients, and Cisco Desktop Collaboration Experience DX Series **Note** CTI route points and CTI ports are covered by calling KW licenses. If your Flex Agreement does not include calling KW licenses, you must obtain separate Enhanced licenses for CTI route points and CTI ports.

Flex 3.0	License Type	Consumed by
NU Professional	CUWL	User and Multiple Devices Cisco Unified IP Phone 7800 Series, Cisco Unified IP Phone 7900 Series, Cisco IP Phones 8811, 8841, 8851, 8861, 8875, Cisco Unified IP Phone 9900 Series, Jabber clients, and Cisco Desktop Collaboration Experience DX Series **Note** CTI route points and CTI ports are covered by calling KW licenses. If your Flex Agreement does not include calling KW licenses, you must obtain separate Enhanced licenses for CTI route points and CTI ports.
EA Access	Basic	Device (low end phones) User and Device (low end phones) Analog phones, Cisco IP Phones 3905, 6901, 7811, 7821, Cisco IP Phone 9811.
EA Common Area	Enhanced	Device User and Device Cisco IP Phones 7841 and 7861, Cisco IP Phones 7945 and 7965, Cisco IP Phones 8811, 8821, 8841, 8851, 8861, 8875, Cisco IP Phones 9841, 9851, 9861, and 9871, Cisco Unified IP Phones 9900 Series, Jabber clients, and Cisco Desktop Collaboration Experience DX Series
EA Knowledge Worker	CUWL	User and Multiple Devices Cisco Unified IP Phone 7800 Series, Cisco Unified IP Phone 7900 Series, Cisco IP Phones 8811, 8841, 8851, 8861, 8875, Cisco Unified IP Phone 9900 Series, Jabber clients, and Cisco Desktop Collaboration Experience DX Series

Flex 3.0	License Type	Consumed by
UC Manager TelePresence Room	TelePresence Room license	Supports room based immersive and multipurpose Cisco TelePresence System endpoints and Spark Room. (For example: Cisco Desk Pro G2, Cisco Codec Pro G2, Cisco Board Pro 55 G2, and Cisco Board Pro 75 G2, Cisco TelePresence System Series 3200, 3000, 1300, Cisco TelePresence MX Series, Cisco TelePresence TX Series, Cisco TelePresence System Profile Series, Cisco Expressway)

License Compliance

When first installed, the Unified Communications Manager is fully operational in demonstration mode for an evaluation period of 90 days, until it has successfully registered with the Cisco Smart Software Manager. After registration, the Unified Communications Manager communicates with Cisco Smart Software Manager periodically. The Unified Communications Manager reports the total license requirements by license type to the Cisco Smart Software Manager and then gets back the license status.

Licenses in the non-compliant state for Unified Communications Manager are enforced after a 90-day overage period. At the conclusion of the grace period, Unified Communications Manager enforces non-compliance with the following service degradation:

Devices and Users cannot be provisioned. Changing the configuration of a user that affects licensing (For example: the Enable/Disable IM and Presence and the Enable/Disable Mobility check boxes) is not allowed.

For information about smart licensing operations, see the System Configuration Guide for Cisco Unified Communications Manager

User Only Licensing

If a user is configured on the system and is not associated with a device, that user does not own any devices and is a "User Only." A user is associated with a device or owns the device if that user's user ID is entered in the OwnerUserID field of the device. The licensing for a "User Only" is shown in the User and Device Support table, for the user not associated with any devices.

Simply adding a user to the system does not consume a license if that user does not own any devices or does not use a licensed user feature. If, however, the user is configured with a licensed user feature, or that user does own a device, then the user does consume a license. The only licensed feature currently is Mobile Connect (also known as Mobility or Single Number Reach or SNR).

Mobile Connect (or Mobility or Single Number Reach) for a user is configured when a Remote Destination Profile (RDP) has been created with the end-user set as the Device owner (User ID field).

Device Only

If a device is added to Cisco Unified Communications Manager and does not have an entry for OwnerUserID field in its Device Configuration window, then the device is not assigned or not associated to a user and called "Device Only". The licensing for "Device Only" devices is listed in the Cisco Unified Communications Manager Licensing - User and Device Support table. If a device is added to Cisco Unified Communications Manager and does not have an entry for OwnerUserID, then the device would require the minimum license type determined by device type, as shown in the Licensing - User and Device Support table.

User and Device

Once a device is assigned or associated with a user, by entering a user ID in the OwnerUserID field of the device, the licensing requirements for that user and device are determined by the type of device and the number of devices assigned to the user. For a user that owns one device, if the user ID of that user is added as OwnerUserID to one Essential device (such as a 3905, 6901, or analog device), then the User and Device minimum license that is required is an Essential license. This means that the one Essential license supports both the user and the device. If instead, that user ID of that user is added as OwnerUserID to one Basic device (such as 6911 or 6921), then the user and device minimum license that is required is one Basic license. If the user ID of a user is added as OwnerUserID to one Enhanced device, the user and device minimum license that is required is an Enhanced license.

For users that own more than one device, the minimum licensing is determined by the number of devices that are owned by the user. Cisco Unified Communications Manager Licensing table shows the maximum number of devices per user license that is supported. A user that owns two devices requires an Enhanced Plus license at a minimum. A user that owns more than two devices requires a CUWL license at a minimum.

Cisco Unified Communications Manager Licensing - User and Device Support table summarizes Cisco Unified Communications Manager Licensing for User Only, Device Only, and User and Device.

Table 15: Cisco Unified Communications Manager Licensing - User and Device Support

License Type	Device Only	User and Device	User Only
UC Manager Essential	• Cisco Unified SIP Phone 3905 • Cisco Unified IP Phone 6901 • Analog devices	A user with 1 Essential device.	N/A
UC Manager Basic	Cisco Unified IP Phone 6911 and 6921 models OR Any device from **UC Manager Essential** license type.	A user with 1 Basic device. OR A user and associated device from the **UC Manager Essential** license type.	A user with Single Number Reach (Mobile Connect). OR A user with an **UC Manager Essential** license type.

License Type	Device Only	User and Device	User Only
UC Manager Enhanced		A user with 1 Enhanced device. OR A user and associated device with **UC Manager Essential** or **UC Manager Basic** license type.	N/A

License Type	Device Only	User and Device	User Only
	• Cisco Unified IP Phone 3911, 3941, 3951 • Cisco Unified IP Phone 6941, 6945, and 6961 models • Cisco Unified IP Phone 7900 Series (7900G, 7911G, 7912G, 7931G, 794xG, 796xG, and 7975G models) • Cisco Unified IP Phone 8900 Series (8941, 8945, and 8961 models) • Cisco Unified IP Phone 9900 Series (9951 and 9971 models) with or without a camera • Cisco Unified Wireless IP Phones Series (792xG and 7925G-EX models) • Cisco Unified IP Conference Stations (7936G and 7937G stations) • Cisco Unified Softphones (Cisco Unified Personal Communicator, Cisco UC Integration for Lync, Cisco UC Integration for Connect, and Cisco IP Communicator) • Jabber clients (Jabber for Mac, Jabber for Windows, Jabber for iPhone, Jabber for Android, Jabber for iPad, and		

License Type	Device Only	User and Device	User Only
	Jabber SDK) • Cisco Virtual Experience Clients (VXC) with voice and video firmware • Cisco IP Video Phone E20 • Cisco TelePresence System EX Series (EX60 and EX90) • Third-party SIP devices • Cisco Desktop Collaboration Experience DX600 Series • Transnova S3 • Cisco Spark Room Device • IMS **Note** CTI route points and CTI ports are covered by calling KW licenses. If your Flex Agreement does not include calling KW licenses, you must obtain separate Enhanced licenses for CTI route points and CTI ports. OR Any device from the UC Manager Essential or UC Manager Basic license type.		

License Type	Device Only	User and Device	User Only
UC Manager Enhanced Plus	N/A	A user with 2 devices. OR A user and associated devices with **UC Manager Essential, UC Manager Basic, UC Manager Enhanced**, or **UC Manager Enhanced Plus** license type.	N/A

License Type	Device Only	User and Device	User Only
UC Manager TelePresence Room License		A user with 1 UC Manager TelePresence Room device associated.	N/A

License Type	Device Only	User and Device	User Only
	• Cisco TelePresence System 500 Series • Cisco TelePresence System 1100 • Cisco TelePresence System 1300 Series • Cisco TelePresence System 3000 Series • Cisco TelePresence System 3200 Series • Cisco TelePresence TX9000 Series (TX9000, TX9200) • Cisco TelePresence TX1300 Series • Cisco TelePresence System Profile Series (42-inch 6000 MXP, 52- inch MXP, 52- inch Dual MXP, 65-inch, and 65- inch Dual) • Cisco TelePresence System Codecs C90/C60/C40 • Cisco TelePresence System Quick Set C20 • Cisco TelePresence MX Series (MX300 and MX200) • Cisco TelePresence 1000 • Cisco TelePresence SX series • Cisco Webex devices • Generic Desktop Video Endpoint • Generic Multiple Screen Room		

License Type	Device Only	User and Device	User Only
	System • Generic Single Screen Room System		

Device Only means a device configured in Cisco Unified Communications Manager that does not have a user association, where the OwnerUserID field is blank.

User and Device means a device configured in Cisco Unified Communications Manager that has a user associated, the OwnerUserID field has a registered userid.

User Only means a user configured in Cisco Unified Communications Manager that does not have any devices associated with the user - whose user id is not found as OwnerUserID for any Cisco Unified Communications Manager devices.

Bold text in the above table indicates that a device is supported through license substitution where an available license of the license type listed may be used to meet lower-level license requirements. This is done in Cisco Smart Software Manager.

Note MGCP FXS ports do not require any license because they are not considered analog phones.

Maximum Number of Devices Per User

The Essential, Basic, and Enhanced licenses support users with one associated device, where the user's id is entered in the OwnerUserId field of one device. The Enhanced Plus license supports users with two associated devices. UWL supports users with three and up to ten associated devices.

TelePresence Room License

Multi-purpose and immersive TelePresence devices are licensed under a separate device license type that is called the TelePresence Room license. The TelePresence Room license covers both the TelePresence device and phone that is registered to Cisco Unified Communications Manager, only if the same userid is entered as the OwnerUserID field for the TelePresence device and the phone. If the same userid is not entered as OwnerUserID for both the TelePresense device and the phone, then the devices are not associated and two licenses are required: one TelePresence Room license for the device and one Enhanced for the phone. The TelePresence touch device does not register to the Cisco Unified Communications Manager, and therefore does not require a separate license or the OwnerUserID association.

License Substitution

The Cisco Smart Software Manager (CSSM) allows for tiered license substitution of available licenses to enable compliance. The available higher-level licenses are substituted or loaned to meet lower level license requirements. For example, if a customer has 100 UC Manager CUWL licenses installed, however Cisco

Unified Communications Manager is reporting back license requirements for 10 CUWL licenses and 50 UC Manager Enhanced Plus licenses, CSSM will calculate that there are 100-10 or 90 UC Manager CUWL license available to be loaned to lower tiers. Of the 90 UC Manager CUWL available licenses 50 CUWL would then be used to meet the requirements for the 50 Enhanced Plus licenses. CSSM will show 40 UC Manager CUWL licenses as available.

Note When Cisco Smart Software Manager On-Prem (Cisco SSM On-Prem) or Smart Software Manager satellite is used in Unified Communications Manager for licensing, there is a difference in the way the license Hierarchy Substitution breakdown is displayed in CSSM, when compared to Cisco SSM On-Prem. See the Cisco SSM On-Prem user interface for details on the insufficient license information if the license authorization status of Unified CM is Out of Compliance. Refer CSCwf47221 for more details.

Note If the Virtual account is already used by Product Instance using direct communication and license reserved for Specific License Reservation, the available license quantities are shown incorrectly. Refer CSCwf47223 for more details.

Licensing Scenarios

The following licensing scenarios will walk through the configuration changes on the Cisco Unified Communications Manager Administration that result in licensing requirements.

Adding Users

When a new user (UserA) is first added to Cisco Unified Communications Manager Administration through the End User configuration or through the Bulk Administration tool, if the user does not have remote device profiles under Enable Mobility, then the new user does not require a license.

If a new user (UserB) is first added to Cisco Unified Communications Manager with remote destination profiles configured under Enable Mobility, then the new user, UserB, requires a Basic license.

UserID	Licensed User Feature	License Required	Note
UserA	None	None	With no assigned devices
UserB	Mobility	Basic	With no assigned devices

Adding Unassociated Devices

If a new device is registered to Cisco Unified Communications Manager and there is no user id entered in the OwnerUserID field for the device, then the device is unassociated to a user, and requires the license per device type for unassociated devices, as indicated in Cisco Unified Communications Manager Licensing - User and

Device Support table. For example, Device6901 is added and it requires an Essential license. Device6921 is added and it requires a Basic license. DeviceEX60 is added and it requires an Enhanced device.

There are currently no devices that require an Enhanced Plus, CUWL Standard, or CUWL Professional license. So you will not see a requirement in Cisco Unified Communications Manager for an unassociated device that requires an Enhanced Plus or above license.

Table 16: Example Device Only License Requirements

Device	License Required	Note
Device6901	UC Manager Essential	With no OwnerUserID
Device6921	UC Manager Basic	With no OwnerUserID
DeviceEX60	UC Manager Enhanced	With no OwnerUserID

Adding Users with Associated Devices

When a device is added, if the device is associated with a user, then the user and device share a license. For one device per user, the license that is required is the greater of the user license or device license required. The following scenarios review the different combinations of device and user associations for one device per user.

Essential Device Associated to User

If Device6901 (an Essential device) is assigned to UserA, by entering OwnerUserID = UserA, then both the device and user are supported by one Essential license.

If however, Device6901 (an Essential device) is assigned to UserB (a Basic user), by entering OwnerUserID = UserB then both device and user are supported by one Basic License.

Basic Device Associated to User

If Device6921 (a Basic device) is assigned to UserA by entering OwnerUserID = UserA, then both the device and user are supported by one Basic license. Similarly, if Device6921 (a Basic device) is assigned to UserB (a Basic user) by entering OwnerUserID = UserB, then both the device and user are supported by one Basic license.

Enhanced Device Associated to User

Most physical phones, soft clients, and desktop video devices such as the EX60 and EX90 are included in the Enhanced device level. If Device EX60 (an Enhanced device) is assigned to UserA by entering OwnerUserID = UserA, then both the device and user is supported by one Enhanced license. Similarly, if DeviceEX60 (an Enhanced device) is assigned to UserB (a Basic user) by entering OwnerUserID = UserB, then both the device and user are supported by one Enhanced license.

Table 17: Example Users and Device License Requirements

Device	OwnerUserID	Licensed User Feature	License Required
Device6901	UserA	None	UC Manager Essential
	UserB	Mobility	UC Manager Basic
Device6921	UserA	None	UC Manager Basic
	UserB	Mobility	UC Manager Basic
DeviceEX60	UserA	None	UC Manager Enhanced
	UserB	Mobility	UC Manager Enhanced

Number of Devices Per User

The above examples for Users and Devices apply only when a user is associated with one device - where their userid is found in only one device configuration OwnerUserID field. When a user is associated with more than one device, then higher level licenses are required independent of device type.

If UserA is assigned to OwnerUserID for one device then the scenarios above apply. If, however, UserA is assigned OwnerUserID for two devices, then one Enhanced Plus license is required for both the user and the two associated devices. If UserA is assigned OwnerUserID for more than two devices, then one UWL Standard license is required. UserA can be assigned up to ten devices with one UWL Standard license. If more than ten devices are assigned to one user, then the user requires one UWL Standard license and also requires an additional license for the additional device.

License Usage Report

Usage details are available by license type, users, and unassigned devices. Usage information is updated once every six hours and maybe updated manually by clicking on Update Usage Details. Clicking Update Usage Details is a resource-intensive process and may take a few minutes depending on the size of your system. There is a link provided to review the Unified Communications licensing information in **View all license type descriptions and device classifications.**

The **Status** message displays if there is an alarm or licensing alert (license non-compliance). See Alarms alerts and license status notifications for further information on status messages. See License Compliance for further information on license compliance and non-compliance.

The **License Requirements by Type** table shows the current system license requirements. It shows current license usage (number of licenses required) by license type and summarizes the number of users and unassigned devices that are requiring licenses by license type. The Report links by license type are provided by (number of) Users or (number of) Unassigned devices and allow drill-down links. For the User report, the user id link provides details on user configuration per the user id. The view details link provides license requirements per user id. For the Unassigned Devices report, the Device Type and License Type that is required is displayed for each unassigned device.

License Usage Reports are also available summarized by Users and Unassigned devices. The Users row lists the total number of users configured on the system. View Usage Report for the users provides a report for all

users configured on the system and their corresponding license requirements. View Usage Report for the Unassigned Devices shows the total number of unassigned devices (devices with no associated user).



Note Assigning a user ID to a device using Cisco Unified Communications Administration moves the device from "Unassigned Devices" to "Users" in the License Usage Report. However, adding a device to the list of controlled devices for an end-user does not modify the "License Usage Report" results for the device.

Cisco Unified Reporting

The following reports are available from the Cisco Unified Reporting console for Cisco Unified Communications Solutions.

1. From Cisco Unified Communications Manager Administration login page Navigation bar, click Cisco Unified Reporting.
2. Choose System Reports.
3. Choose Unified CM Device Counts Summary.

The generated report will summarize, per cluster, the device counts by model.

1. From Cisco Unified Communications Manager Administration login page Navigation bar, click Cisco Unified Reporting.
2. Choose System Reports.
3. Choose Unified CM User Device Count.

The generated report will summarize, per cluster, the phone to user relationship the number of phones with no users, users with one phone, and users with more than one phone.

1. From Cisco Unified Communications Manager Administration login page Navigation bar, click Cisco Unified Reporting.
2. Choose System Reports.
3. Choose Unified CM User Device Count.

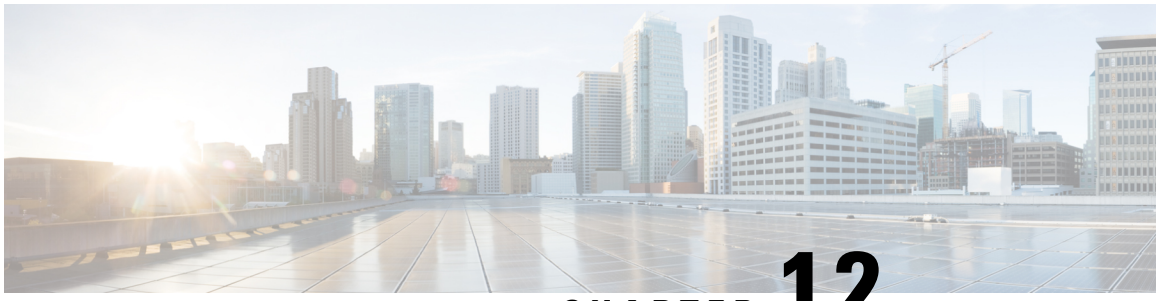
The generated report will summarize, per cluster, the phone to user relationship the number of phones with no users, users with one phone, and users with more than one phone.



PART V

Monitoring and Recording

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CHAPTER 12

Silent Monitoring

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- Silent Monitoring Prerequisites, on page 122
- Configure Silent Monitoring Task Flow, on page 122
- Silent Monitoring Interactions, on page 127
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Silent Monitoring Overview

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

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

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

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

Secure Silent Monitoring

You can also configure secure silent monitoring. Secure silent monitoring allows encrypted media (sRTP) calls to be monitored. Monitoring calls are always established using the highest level of security that is determined by the capabilities of the agent phone regardless of the security status of the call being observed. The highest level of security is maintained by exchanging the secure media key in any call between the customer, agent, and supervisor. Monitoring calls using secured media carries approximately 4000 bits per second of additional bandwidth overhead, same as standard secure media (sRTP) calls.

If the agent phone has encryption that is enabled, the supervisor phone must also have encryption enabled in order to allow secure silent monitoring. If the agent phone has encryption that is enabled, but the supervisor phone does not, the monitoring request fails.

Whisper Coaching

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

Silent Monitoring Prerequisites

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

- Cisco Unified Contact Center Enterprise—For details on how to set up silent monitoring in Cisco Unified Contact Center Enterprise, see Cisco Remote Silent Monitoring Installation and Administration Guide.
- Cisco Unified Contact Center Express—This chapter contains a sample configuration to set up Silent Monitoring for Unified Contact Center Express via Cisco Finesse. For additional documentation that is related to your Cisco Unified Contact Center Express deployment, go to https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-express/tsd-products-support-series-home.html.

Configure Silent Monitoring Task Flow

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

Before you begin

- Determine which phones support silent monitoring by running a phone feature list report. For more information,Generate a Phone Feature List, on page 5

Procedure

	Command or Action	Purpose
Step 1	Perform one of the following procedures: • Enable Built in Bridge for Phones Clusterwide, on page 123 • Enable Built in Bridge for a Phone, on page 123	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.

	Command or Action	Purpose
Step 2	Enable Monitoring Privileges for Supervisor, on page 124	Add the supervisor to a group that allows silent monitoring.
Step 3	Assign a Monitoring Calling Search Space, on page 124	Set up the monitoring calling search space for the supervisor phone.
Step 4	Configure Silent Monitoring Notification Tones, on page 125	Configure whether you want to play notification tones to the call participants.
Step 5	Configure Secure Silent Monitoring, on page 125	**Optional**. If your calls are encrypted, configure secure silent monitoring.
Step 6	Configure Silent Monitoring for Unified Contact Center Express, on page 126	For Unified Contact Center Express deployments, configure Silent Monitoring via Cisco Finesse.

Enable Built in Bridge for Phones Clusterwide

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

Procedure

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

Enable Built in Bridge for a Phone

Use this procedure to enable the Built in Bridge 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 123.

Procedure

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

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

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

Step 4 Click **Save**.

Enable Monitoring Privileges for Supervisor

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

Before you begin

Perform one of the following procedures to enable the Built in Bridge on agent phones:

- Enable Built in Bridge for Phones Clusterwide, on page 123
- Enable Built in Bridge for a Phone, on page 123

Procedure

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

Step 2 Select the supervisor from the list of users.

Step 3 In the **Permissions Information** section, click **Add to Access Control Group**.

Step 4 Add the **Standard CTI Allow Call Monitoring** and **Standard CTI Enabled** user groups.

Step 5 Click **Save**.

Assign a Monitoring Calling Search Space

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

Procedure

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

Step 2 Click **Find** and select the supervisor phone.
The left navigation pane displays the available phone lines for the supervisor's phone.

Step 3 Perform the following steps for each of the supervisor's phone lines that are used for monitoring:

a) Click the phone line. The **Directory Number Configuration** window displays configuration information for that phone line.

b) From the **Monitoring Calling Search Space** drop-down list, choose a calling search space that includes both the supervisor phone line and the agent phone line.
c) Click **Save**.

Configure Silent Monitoring Notification Tones

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Service Parameters**.
Step 2 From the **Server** drop-down list, choose the server one which the CallManager service is running.
Step 3 From the **Service** drop-down list, choose **Cisco CallManager**.
Step 4 Configure values for the following service parameters:

- If you want to play a notification tone to the agent, change the value of the **Play Monitoring Notification Tone To Observed Target** service parameter to **True**.
- If you want to play a notification tone to the customer, change the value of the **Play Monitoring Notification Tone To Observed Connected Parties** service parameter to **True**.

Step 5 Click **Save**.
Step 6 Reset the agent phone, if you changed the service parameter configuration.

Configure Secure Silent Monitoring

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

Procedure

	Command or Action	Purpose
Step 1	Configure an Encrypted Phone Security Profile , on page 126	Configure phone security profiles that include encryption for the agent phone and supervisor phone.
Step 2	Assign Security Profile to Phone, on page 126	Apply the encrypted phone security profile to the agent phone and the supervisor phone.

Configure an Encrypted Phone Security Profile

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

Procedure

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

Step 2 Perform either of the following steps:

- Click **Add New** to create a new phone security profile.
- Click **Find** and select an existing phone security profile.

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

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

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

Step 6 Click **Save**.

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

Assign Security Profile to Phone

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

Procedure

Step 1 From Cisco Unified CM Administration, chooseI **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, choose the phone security profile that you have set up.

Step 4 Click **Save**.

Step 5 Repeat the previous steps for the supervisor phone.

Configure Silent Monitoring for Unified Contact Center Express

The following steps contain a sample Silent Monitoring for Cisco Unified Contact Center Express configuration via Cisco Finesse.

Before you begin

Make sure that both the agent and supervisor phone are compatibile for Cisco Finesse. Refer to the *Unified CCX Software Compatibility Matrix* at https://www.cisco.com/c/en/us/support/customer-collaboration/unified-contact-center-express/products-device-support-tables-list.html.

Procedure

Step 1 Configure a test agent and supervisor on Unified Contact Center Express.

Note
The IP Contact Center (IPCC) Extension for the Agents and Supervisors must be unique. This can be verified from Cisco Unified Communications Manager under **Call Routing** > **Route Plan Report**.

Step 2 Ensure that the agent phone has the Built in Bridge (BIB) On. This can be done on the phone or at the Cluster level (Set the Default Service Parameter to On).

Step 3 Log in to Finesse as an Agent.

Step 4 Log in to Finesse as a Supervisor and ensure that the supervisor is in NOT READY State.

Step 5 Ensure that the Resource Manager Contact Manager (RMCM) user has the required roles for Call Monitoring and Call Recording -- Standard Computer Telephony Integration (CTI) Allow Call Monitoring and Recording.

Note
This is automatically done by Unified Contact Center Express at the initial setup of the RMCM user. Ensure the roles exist on the **Application User** window of Cisco Unified Communications Manager.

Step 6 Assign the Monitoring CSS (Calling Search Space) on the Supervisor Phone to contain the Partition of the agent line.

Step 7 Place a call to Unified Contact Center Express so that the call is routed to the agent logged in. Once the agent is in the TALKING state, from the supervisor, start the Silent Monitoring. The supervisor will then be able to hear the conversation between the agent and the caller

Silent Monitoring Interactions

Feature	Interaction
Call preservation	If the agent call that is being monitored goes to call preservation, Unified Communications Manager also puts the monitoring call into call preservation mode.
Transfer of secure monitoring call	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.
Recording Tones	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.

Feature	Interaction
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

Feature	Restriction
Barge	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.
Transfer of Secure Silent Monitoring over an intercluster trunk	Unified Communications Manager does not support transferring Secure Silent Monitoring calls over an intercluster trunk.
Silent Monitoring Restriction	The monitoring fails if the supervisor logs in through the non-secure mode and the Agent logs in to the MRA mode. For more information, see the *Secure Silent Monitoring*section.



CHAPTER 13

Recording

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- Recording Prerequisites, on page 132
- Recording Configuration Task Flow, on page 133
- Recording Call Flow Examples, on page 142
- Recording Interactions and Restrictions, on page 142

Recording Overview

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

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

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

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

- **Automatic Silent Recording**—Automatic silent recording records all calls on a line appearance automatically. Unified Communications Manager invokes the recording session automatically with no visual indication on the phone that an active recording session is established.
- **Selective Silent Recording**—A supervisor can start or stop the recording session via CTI-enabled desktop. Alternatively, a recording server can invoke the session based on predefined business rules and events. There is no visual indication on the phone that an active recording session is established.
- **Selective User Call Recording**—An agent can choose which calls to record. The agent invokes the recording session through CTI-enabled desktop, or by a softkey or programmable line key. When selective user call recording is used, the Cisco IP phone displays recording session status messages.

Unified Communications Manager supports recording to a single recording server and can be deployed with CUBE as media proxy to record to more than one recording server.

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

Multi-Fork Recording

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

The multi-forking feature provides the following benefits:

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

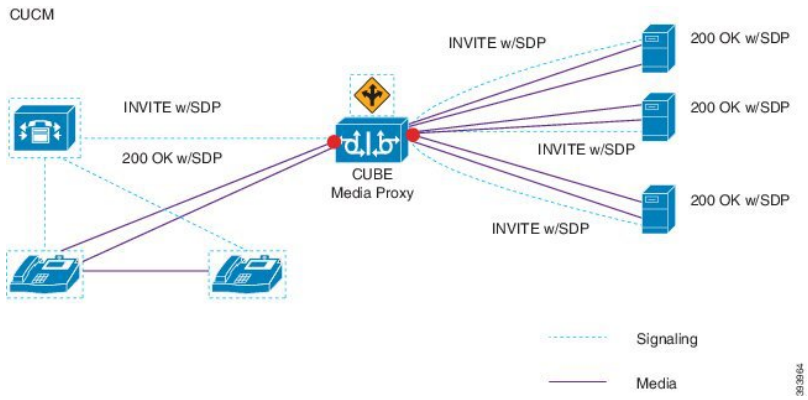
When you implement multi-fork recording, you must configure the CUBE Media Proxy server in your network which connects to Unified Communications Manager over a SIP trunk.

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

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

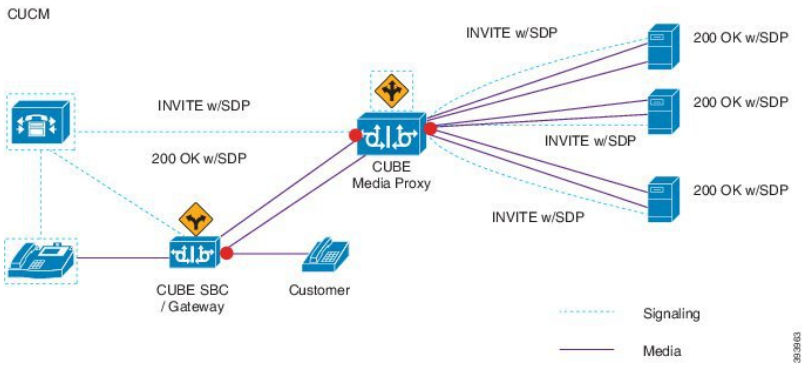
The following example illustrates the phone-based recording of multi-fork recording through CUBE Media Proxy.

Figure 4: Phone-based recording



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

Figure 5: Network-based recording



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

Supported Platforms

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

- Cisco 4000 Series-Integrated Services Routers (ISR G3 - ISR4331, ISR4351, ISR4431, ISR4451).
- Cisco Aggregated Services Routers (ASR - ASR1001-X, ASR1002-X, ASR1004 with RP2, ASR1006 with RP2).
- Cisco Cloud Services Routers (CSR1000V series).

Restrictions for Multi-fork recording through CUBE Media Proxy

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

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

Recording Media Source Selection

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

Table 18: Recording Media Source Selection

Preferred Media Source	Media Type	Gateway in call path?	Selected Media Source
Gateway	Unsecure (RTP)	Yes	Gateway
		No	Phone
	Secure (sRTP)	Yes	Phone
		No	Phone
Phone	Unsecure (RTP)	Yes	Phone
		No	Phone
	Secure (sRTP)	Yes	Phone
		No	Phone

Alternate Recording Media Source if the First Choice is Unavailable

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

Table 19: Alternate Recording Media Source if First Choice is Unavailable

Selected Media Source	Gateway Preferred	Phone Preferred
First attempt	First gateway in call path	Phone
Second attempt	Last gateway in call path	First gateway in call path
Third attempt	Phone	Last gateway in call path

Recording Prerequisites

- Cisco Unified IP Phone support—To view a list of the Cisco Unified IP Phone that support recording, log in to Cisco Unified Reporting and run the Unified CM Phone Feature List report, selecting **Record** as the feature. For a detailed procedure, see Generate a Phone Feature List, on page 5.
- Gateway support—For details on which gateways support recording, see https://developer.cisco.com/web/sip/wiki/-/wiki/Main/Unified+CM+Recording+Gateway+Requirements.
- If you are configuring multiple-stream recording, deploy and configure a CUBE Media Proxy. For details, see the section *CUBE Media Proxy* in the Cisco Unified Border Element Configuration Guide.

Recording Configuration Task Flow

Before you begin

Procedure

	Command or Action	Purpose
Step 1	Create a Recording Profile, on page 133	Create a recording profile.
Step 2	Configure SIP Profile for Recording, on page 134	**Optional**. Configure the SIP Profile if you want to deliver the Conference Bridge Identifier to the recorder.
Step 3	Configure SIP Trunks for Recording, on page 134	Configure the recorder server or CUBE Media Proxy as a SIP trunk device.
Step 4	Configure Route Pattern for Recording, on page 135	Create a route pattern that routes to the recorder server or CUBE Media Proxy.
Step 5	Configure Agent Phone Line for Recording, on page 135	Configure the agent phone line for recording.
Step 6	Enable the built in bridge for your agent phones. Perform one of the following tasks to enable the built-in-bridge for recording: • Enable Built in Bridge for Cluster , on page 136 • Enable Built in Bridge for a Phone, on page 136	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. **Note** The Built in Bridge setting on individual phones overrides the clusterwide defaults.
Step 7	Enable Gateway for Recording, on page 137	Configure Unified Communications services on the gateway.
Step 8	Configure Recording Notification Tones, on page 138	Configure whether you want a notification tone to play when calls are recorded.
Step 9	Perform one of the following procedures, depending on whether your phone uses feature buttons or softkeys: • Configure a Record Feature Button, on page 138 • Configure a Record Softkey, on page 139	Configure a Record feature button or softkey for your phone.

Create a Recording Profile

Use this procedure to create a recording profile.

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name for your recording profile.

Step 4 In the **Recording Calling Search Space** field, select the calling search space that contains the partition with the route pattern that is configured for the recording server.

Step 5 In the **Recording Destination Address** field, enter the directory number or the URL of the recording server or the URL of the CUBE Media Proxy server.

Step 6 Click **Save**.

Configure SIP Profile for Recording

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

Procedure

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

Step 2 Select the SIP profile that you want to use for your network.

Step 3 Set a value for the **Early Offer Support for Voice and Video** calls field. SIP trunk from Unified Communications Manager to CUBE Media Proxy server must be enabled for an Early Offer support and configuration options are **Best Effort (no MTP inserted)** and **Mandatory (insert MTP if needed)**.

Note
We recommend that you enable SIP trunk for Mandatory (insert MTP if needed).

Step 4 Check the **Deliver Conference Bridge Identifier** check box.

Step 5 Click **Save**.

Configure SIP Trunks for Recording

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

Procedure

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

Step 2 Click **Add New**.

Step 3 From the **Trunk Type** drop-down list, choose **SIP Trunk**.

Device Protocol is auto-populated to **SIP**, which is the only available option.

Step 4 From the **Trunk Service Type** drop-down list, choose the service type that you want to use in your network. The default value is **None**.

Step 5 Click **Next**.

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

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

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

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

Step 9 Click **Save**.

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

Configure Route Pattern for Recording

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

Procedure

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

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

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

Step 4 For call recording, complete the following fields:

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

Step 5 Click **Save**.

Configure Agent Phone Line for Recording

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

Procedure

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

Step 2 Click **Find**.

Step 3 Select the agent's phone.

Step 4 In the left Association pane, click the phone line for the agent to view the settings.

Step 5 From the **Recording Option** drop-down list, choose one of the following options:

- **Call Recording Disabled**—Calls on this phone line are not recorded.
- **Automatic Call Recording Enabled**—All calls on this phone line are recorded.
- **Selective Call Recording Enabled**—Only selected calls on this phone line are recorded.

Step 6 From the **Recording Profile** drop-down list, choose the recording profile that is configured for the agent.

Step 7 From the **Recording Media Source** drop-down list, choose whether you want to use the gateway or the phone as the preferred source of recording media.

Step 8 Set the **Busy Trigger** field to a minimum of **3** if you also have Multilevel Precedence and Preemption (MLPP) configured.

Step 9 Click **Save**.

Enable Built in Bridge for Cluster

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

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

Procedure

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

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

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

Step 4 Set the **Builtin Bridge Enable** service parameter to **On**.

Step 5 Click **Save**.

Enable Built in Bridge for a Phone

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

Optionally, use a service parameter to set the Built in Bridge defaults across the cluster. For more information, see Enable Built in Bridge for Cluster , on page 136.

Procedure

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

Step 2 Click **Find** to select the agent phone.

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

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

Note
The recording can fail if the **Built-in-Bridge** is ON and if you check the **Media Termination Point Required** check box.

Step 4 Click **Save**.

Enable Gateway for Recording

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

Procedure

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

Step 2 Configure the XMF Provider.

Step 3 Verify Unified Communications Gateway Services.

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

- For more information, see ASR routers Cisco Unified Border Element (Enterprise) Protocol-Independent Features and Setup Configuration Guide. Cisco IOS XE Release 35.
- For more information, see ISR routers Cisco Unified Border Element Protocol-Independent Features and Setup Configuration Guide, Cisco IOS Release 15M&T.

Configure Recording Notification Tones

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

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

Procedure

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

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

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

Step 4 If you want a notification tone to be played to the agent, set the **Play Recording Notification Tone to Observed Target (agent)** service parameter to **True**.

Step 5 If you want a notification tone to be played to the customer, set the value of the **Play Recording Notification Tone To Observed Connected Parties (customer)** service parameter to **True**.

Step 6 Click **Save**.

Configure a Record Feature Button

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

Procedure

	Command or Action	Purpose
Step 1	Configure a Phone Button Template for Recording, on page 138	Configure a phone button template that includes the Record button.
Step 2	Associate a Phone Button Template with a Phone, on page 139	Associate the phone button template that you set up for recording to the phone.

Configure a Phone Button Template for Recording

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

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate a Phone Button Template with a Phone

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

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure a Record Softkey

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

Procedure

	Command or Action	Purpose
Step 1	Configure a Softkey Template for Recording, on page 140	Configure a softkey template that includes the Record softkey.

	Command or Action	Purpose
Step 2	Perform one of the following procedure: • Associate a Softkey Template with a Phone, on page 141 • Associate a Softkey Template with a Common Device Configuration, on page 141	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.

Configure a Softkey Template for Recording

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.

- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

Associate a Softkey Template with a Phone

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

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Associate a Softkey Template with a Common Device Configuration

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

Procedure

	Command or Action	Purpose
Step 1	Add a Softkey Template to the Common Device Configuration, on page 141	
Step 2	Add Common Device Configuration to Phone, on page 142	

Add a Softkey Template to the Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Add Common Device Configuration to Phone

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

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 Call Flow Examples

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

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/configExamples/cucm_b_recording-use-cases.html

Recording Interactions and Restrictions

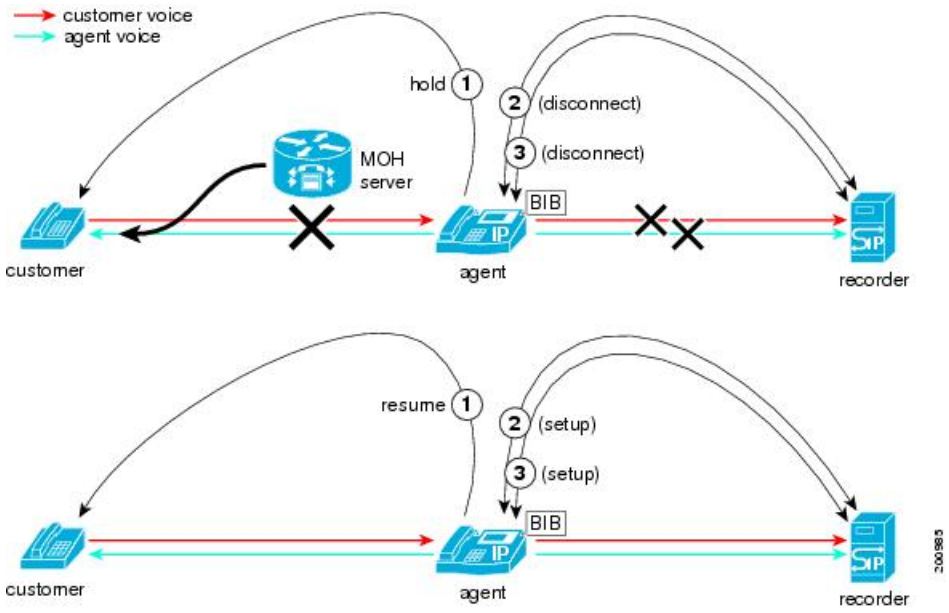
Feature	Interactions and Restrictions
Monitoring Tones	Recording Tones take precedence over Monitoring Tones for calls that are both recorded and monitored. If you configure both, and a call is both recorded and monitored, only the recording tone plays.
Multilevel Precedence and Preemption	If you also have Multilevel Precedence and Preemption (MLPP) configured, you must set the **Busy Trigger** setting on the agent phone line to a minimum of 3.

Feature	Interactions and Restrictions
Secure Tones	If you configure Secure Tones, the secure tone plays to both call participants at the outset of a secure call, irrespective of whether Recording Tones are configured. If you configure both Secure Tones and Recording Tones and the call is secure, the secure tone plays once at the outset of the call followed by the recording tone. If you configure Secure Tones, Recording Tones, and Monitoring Tones, and the call is secured, recorded, and monitored, the secure tone plays once followed by the recording tone. The monitoring tone does not play.
Customer Voice Portal	Agent - customer calls that are routed through the Customer Voice Portal may be recorded using the agent phone as the recording source.
SIP Proxy Servers	If you are using the gateway as your recording source, you cannot place SIP proxy servers between Unified Communications Manager and the gateway.
Busy Hour Call Completion Rate	Each recording session adds two calls to the Busy Hour Call Completion (BHCC) rate with a minimal impact on CTI resources.
Selective Recording with Media Sense	When you configure Selective Recording, the Media Sense server does not record the consult call during a transfer. For example, if a call between an agent and a customer is being recorded and the agent initiates a transfer to a second agent, the consult call that takes place between the two agents, before the call being transferred, is not recorded. To ensure that the consult call is recorded, the agent must press the 'Record' softkey when the consult call starts.
Recording on authenticated phones	To record a call for authenticated phones on the Cisco Unified CM Service Parameter page, set the **Authenticated Phone Recording** field to **Allow Recording**. The default value is **Do Not Allow Recording**. Unified Communications Manager allows call recording for authenticated phones while using a non secure recorder. In case of secure recorder, recording is allowed only if the recorder supports a Secure Real-Time Transport protocol (SRTP) fallback.
Codec locking for auto recording calls in select and join conference	When recording is enabled, the endpoint advertises a single codec and payload, and a select and join conference takes place in the Unified Communications Manager.

Recording Calls Do Not Survive Agent Hold

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

Figure 6: Recording Calls Do Not Survive Agent Hold

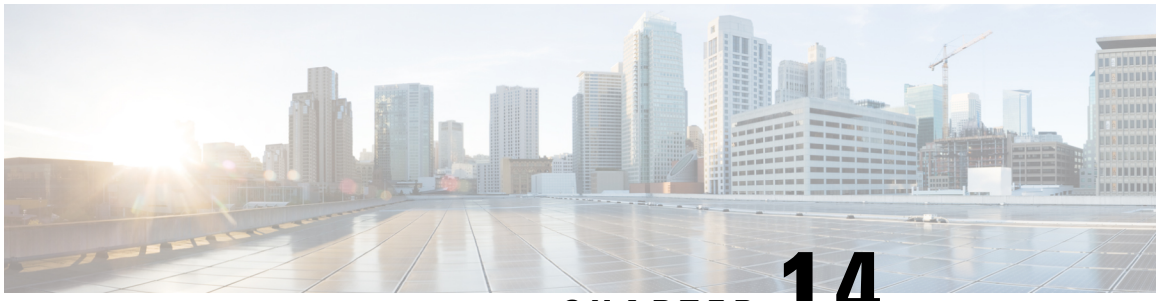




PART VI

Call Center Features

- Agent Greeting , on page 147
- Auto-Attendant , on page 151
- Manager Assistant , on page 159



CHAPTER 14

Agent Greeting

- Agent Greeting Overview, on page 147
- Agent Greeting Prerequisites, on page 147
- Agent Greeting Configuration Task Flow, on page 147
- Agent Greeting Troubleshooting, on page 149

Agent Greeting Overview

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

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

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

Agent Greeting Prerequisites

- Install Cisco Unified Contact Center Enterprise. See Cisco Unified Contact Center Enterprise Installation and Upgrade Guide.
- Install Cisco Unified Customer Voice Portal. See Installation and Upgrade Guide for Cisco Unified Customer Voice Portal .
- Ensure that you enable Built In Bridge. To view the details, see Configure Built In Bridge, on page 149.

Agent Greeting Configuration Task Flow

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

Before you begin

- Review Agent Greeting Prerequisites, on page 147

Procedure

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

Configure Built In Bridge

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

Procedure

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

Step 2 Click **Find** to select the agent phone.

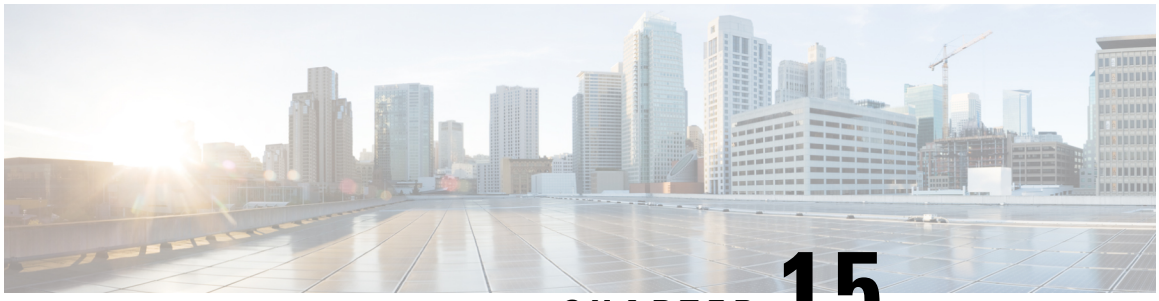
Step 3 From the **Built in Bridge** drop-down list, choose one of the following options:

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

Step 4 Click **Save**.

Agent Greeting Troubleshooting

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



CHAPTER 15

Auto-Attendant

- Auto-Attendant Overview, on page 151
- Cisco Unity Connection Configuration, on page 152
- Cisco Unified CCX Configuration, on page 156
- Cisco Unity Express Configuration, on page 158

Auto-Attendant Overview

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

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

Auto-Attendant provides the following functions:

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

    If the caller chooses to spell by name (by pressing 2), the system compares the letters that are entered with the names that are configured to the available extensions. One of the following results can occur:
    - If a match exists, the system announces a transfer to the matched user and waits for up to 2 seconds for the caller to press any Dual Tone Multifrequency (DTMF) key to stop the transfer. If the caller does not stop the transfer, the system performs an explicit confirmation: it prompts the user for confirmation of the name and transfers the call to the primary extension of that user.
    - If more than one match occurs, the system prompts the caller to choose the correct extension.
    - If too many matches occur, the system prompts the caller to enter more characters.

- If no match occurs, that is, if the user presses wrong options, the system prompts that the user pressed the wrong options and prompts the user to press the correct options.

- When the caller specifies the destination, the system transfers the call.
- If the line is busy or not in service, the system informs the caller accordingly and replays the main menu prompt.

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

- Auto-Attendant using Cisco Unity Connection (CUC); the most widely used Auto-Attendant solution configuration by customers
- Auto-Attendant using Cisco Unified Contact Center Express (Unified CCX)
- Auto-Attendant using Cisco Unity Express (CUE)

Cisco Unity Connection Configuration

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

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

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

Cisco Unity Connection Configuration Task Flow

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

Procedure

	Command or Action	Purpose
Step 1	Configure CTI Route Point, on page 153	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).
Step 2	Configure Auto-Attendant System Call Handler, on page 154	Call handlers answer calls, greet callers with recorded prompts, provide callers with information and options, route calls, and take messages. **Note**

	Command or Action	Purpose
		You can customize the greeting for the AutoAttendant Call Handler by choosing **Edit** > **Greetings**. For more information about customizing greetings, see System Administration Guide for Cisco Unity Connection.
Step 3	Configure Caller Input Option, on page 154	Caller input option enables you to designate a single digit to represent a user extension, alternate contact number, call handler, interview handler, or directory handler. The caller presses a single key during a call handler greeting instead of entering the full extension, and Cisco Unity Connection responds accordingly. Several different keys configured as caller input options offers the callers a menu of choices in the call handler greeting.
Step 4	Configure Extension for Operator Call Handler, on page 155	Configure an extension for the operator to allow callers to speak to an operator during a call handler greeting.
Step 5	Modify Standard Call Transfer Rule for Operator, on page 155	Modify the Standard Call Transfer Rule to enable the call to be transferred to the operator when the caller presses 0 to speak to an operator.
Step 6	Update Default System Transfer Restriction Table, on page 155	Update the Default System Transfer restriction table. The Default System Transfer restriction table restricts numbers that can be used for Caller system transfers, which allow unidentified callers to transfer to a number that they specify.

Configure CTI Route Point

Procedure

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

Step 2 Click **Add New**.

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

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

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

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

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

Step 8 From the **Route Partition** drop-down list, choose the required route partition.

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

Procedure

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

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

Step 3 In the **Display Name** field, enter **AutoAttendant**.

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

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

Step 6 Edit the required fields and click **Save**.

Configure Caller Input Option

Procedure

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

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

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

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

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

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

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

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

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

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

Configure Extension for Operator Call Handler

Procedure

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

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

Step 3 Enter the extension of the operator in the **Extension** field and click **Save**.
The `Updated Caller Input` status message is displayed.

Modify Standard Call Transfer Rule for Operator

Procedure

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

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

Step 3 From the **Edit** menu, choose **Transfer Rules**.
The **Transfer Rules** window is displayed.

Step 4 Click **Standard**.
The **Edit Transfer Rule (Standard)** window is displayed.

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

Step 6 Click **Save**.

Update Default System Transfer Restriction Table

Procedure

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

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

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

Step 4 Click **Save**.

Cisco Unity Connection Auto-Attendant Troubleshooting

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

- http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/107517-calltrf.html
- http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/troubleshooting/guide/8xcuctsgx/8xcuctsg110.html
- http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/troubleshooting/guide/8xcuctsgx/8xcuctsg040.html
- http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/troubleshooting/guide/8xcuctsgx/8xcuctsg180.html

Cisco Unified CCX Configuration

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

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

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

Cisco Unified CCX Prerequisites

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

Cisco Unified CCX Auto-Attendant Task Flow

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

Before you begin

- Learn more about the Auto-Attendant feature by reviewing Auto-Attendant Overview, on page 151.
- Learn more about Cisco UCCX with Auto-Attendant functionality by reviewing Cisco Unified CCX Configuration, on page 156
- Review Cisco Unified CCX Prerequisites, on page 156.

Procedure

	Command or Action	Purpose
Step 1	Configure Unified CM Telephony call control groups.	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.
Step 2	Add a Cisco Media Termination (CMT) dialog control group.	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 “auto” and contain the same ID as the call control group—not the CMT dialog group. Perform this procedure if the default media termination is configured and the ID differs.
Step 3	Configure a Cisco script application.	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.
Step 4	Provision a Unified CM Telephony trigger.	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.

	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 Unity Express Configuration

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

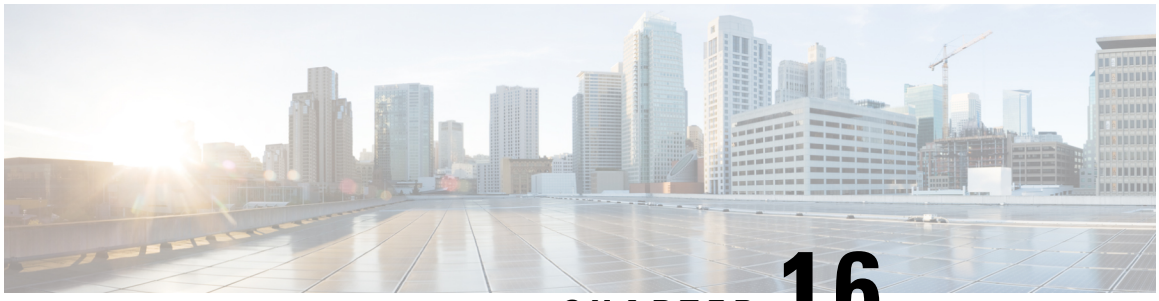
For information about deploying a sample Auto-Attendant script, see "Deployment of sample script aa.aef" chapter in the Getting Started with Cisco Unified IP IVR.

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

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

Cisco Unity Express Auto-Attendant Troubleshooting

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



CHAPTER 16

Manager Assistant

- Cisco Unified Communications Manager Assistant Overview, on page 159
- Manager Assistant Prerequisites, on page 161
- Manager Assistant Task Flow for Proxy Lines, on page 162
- Manager Assistant Task Flow for Shared Lines, on page 170
- Manager Assistant Interactions, on page 188
- Manager Assistant Restrictions, on page 190
- Cisco Unified Communications Manager Assistant Troubleshooting , on page 191

Cisco Unified Communications Manager Assistant Overview

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

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

Manager Assistant supports shared line support and proxy line support.

Manager Assistant Architecture

The Manager Assistant architecture comprises the following:

- **Cisco IP Manager Assistant service**—After you install Unified Communications Manager, activate this service from the Cisco Unified Serviceability interface.
- **Assistant Console interface**—Allows assistants to access the Manager Assistant features on their computer to handle calls for managers. The Manager Assistant handles calls for an assistant and for as many as 33 managers.
- **Cisco Unified IP Phone interface:** Managers and assistants use softkeys and the Cisco Unified IP Phone Services button to access the Manager Assistant features.

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

Manager Assistant Database Access Architecture

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

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



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

- **Assistant interface**—The assistant accesses the Manager Assistant features by using the Assistant Console application and the Cisco Unified IP Phone. The Assistant Console, an application, provides call-control functions such as answer, divert, transfer, and hold. The assistant uses the Assistant Console to log in and log out, to set up assistant preferences, and to display the **Manager Configuration** window that is used to configure manager preferences.

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

Softkeys

Manager Assistant supports the following softkeys:

- Redirect
- Transfer to VoiceMail
- Do Not Disturb

Manager Assistant supports the following softkey templates:

- Standard Manager—Supports manager for proxy mode
- Standard Shared Mode Manager—Supports manager for shared mode
- Standard Assistant—Supports assistant in proxy or shared mode
- Standard User—The system makes call-processing (such as Hold and Dial) softkeys available with the Standard User template.

Manager Assistant Shared Line Overview

When you configure Manager Assistant in shared line mode, the manager and assistant share a directory number, for example, 8001. The assistant handles calls for a manager on the shared directory number. When a manager receives a call on 8001, both the manager phone and the assistant phone ring.

The Manager Assistant features that do not apply to shared line mode include Default Assistant Selection, Assistant Watch, Call Filtering, and Divert All Calls. An assistant cannot see or access these features on the Assistant Console application.

Manager Assistant Proxy Line Overview

When you configure Manager Assistant in proxy line mode, the assistant handles calls for a manager using a proxy number. The proxy number is not the directory number for the manager, but is an alternate number chosen by the system that an assistant uses to handle manager calls. In proxy line mode, a manager and an assistant have access to all features that are available in Manager Assistant, which include Default Assistant Selection, Assistant Watch, Call Filtering, and Divert All Calls.

Manager Assistant Prerequisites

- The user must install JRE on a 32 or 64-bit Windows platform before the user upgrades the Manager Assistant client to a newer version for Releases 11.5(1)SU9, 12.0(1)SU4, and 14 onwards.



Important Before you perform the upgrade, ensure that you uninstall the Cisco Unified Communications Manager Assistant client that is currently installed on your machine. This is applicable from Releases 12.0(1)SU4 and 14 onwards.

- For the Windows 11 platform, the user must install JRE 1.8 on the 64-bit Windows platform before the user upgrades the Manager Assistant client to a newer version for Release 15SU2 and later.
- Manager Assistant supports the following browsers and platform:
  - Unified Communications Manager Assistant Administration and the Assistant Console are supported on Internet Explorer 11 with Windows 10 and 11 (64 bit), Firefox with Windows 10 and 11 (64 bit) or later, and Safari with MacOS (10.x) or later.



Note To run IPMA plug-in on Windows 11 for Unified Communications Manager Release 15SU1 and earlier versions, you should install the IPMA Release 15 version plug-in to any of these supported OS platforms: Windows 10, Windows 2019, and Windows 2022. You must then copy the installed version of the IPMA plug-in to Windows 11 and then launch IMPA.

  - On a computer running Windows 10 and 11 or Apple MAC OS X, you 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 10 and 11, Windows 2019, and Windows 2022.
- 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 *Bulk Administration Guide*.

Manager Assistant Task Flow for Proxy Lines

Before you begin

- Review Manager Assistant Prerequisites, on page 161.

Procedure

	Command or Action	Purpose
Step 1	Run the Cisco Unified CM Assistant Configuration Wizard, on page 162	
Step 2	Configure Manager And Assign Assistant For Proxy Line, on page 168	
Step 3	Configure Assistant Line Appearances for Proxy Line, on page 169	
Step 4	Install Assistant Console Plugin, on page 187	The assistant accesses the 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 5	Configure the manager and Assistant Console applications.	See Cisco Unified Communications Manager Assistant User Guide for Cisco Unified Communications Manager.

Run the Cisco Unified CM Assistant Configuration Wizard

You can run the Cisco Unified CM Assistant Configuration Wizard to automatically create partitions, calling search spaces, and route points. The wizard also creates Bulk Administration Tool (BAT) templates for the manager phones, the assistant phones, and all other user phones. You can use the BAT templates to configure the managers, assistants, and all other users. For more information about BAT, see Bulk Administration Guide for Cisco Unified Communications Manager .

Before you begin

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **Application** > **Cisco Unified CM Assistant Configuration Wizard**.

Step 2 Click **Next** to begin the Cisco Unified CM Assistant Configuration wizard process.

Step 3 In the **Partition for Managers** window, enter a name, provide a description, and then click **Next**. Alternatively, you can accept the default partition name and description.

Step 4 In the **Partition for CTI Route Point** window, enter a name, provide a description, and then click **Next**. Alternatively, you can accept the default CTI route point name.

Step 5 In the **Partition for All Users** window, enter a name, provide a description and then click **Next**. Alternatively, you can accept the default partition name and description for all users.

Step 6 In the **Intercom Partition** window, enter a name, provide a description, and then click **Next**. Alternatively, you can accept the default intercom partition name.

Step 7 In the **Assistant Calling Search Space** window, enter a name, and provide a description. Alternatively, you can use the default calling search space name and description.
The Available Partitions and Selected Partitions boxes under the Route Partitions for this Calling Search Space automatically list Partitions for the Assistant Calling Search Space. You can accept the default values or you can choose the applicable partition from the Available Partitions box. Use the up and down arrows to move partitions from one box to the other.

Step 8 Click **Next**.

Step 9 In the **Everyone Calling Search Space** window, enter a name, and provide a description. Alternatively, you can accept the default calling search space name and description for everyone.
The Available Partitions and Selected Partitions boxes under the Route Partitions for this Calling Search Space automatically list Partitions for the Assistant Calling Search Space. You can accept the default values or you can choose the applicable partition from the Available Partitions box. Use the up and down arrows to move partitions from one box to the other.

Step 10 Click **Next**.
If you have existing calling search spaces that are configured on the system, the **Existing Calling Search Spaces** window is displayed; otherwise, proceed to the next step.

Manager Assistant requires that the existing calling search spaces add the prefix **Generated_Route Point** and **Generated_Everyone** partitions. The Available Calling Search Spaces and Selected Calling Search Spaces boxes automatically list these partitions. Use the up and down arrows to move partitions from one box to the other.

Note
The prefix that is added to the existing calling search spaces may change if the administrator has changed the names of the partitions.

Step 11 Click **Next**.

Step 12 In the **CTI Route Point** window, enter a name in the CTI route point name field; otherwise, use the default CTI route point name.

Step 13 From the drop-down list, choose the appropriate device pool.

Step 14 Enter a route point directory number; otherwise, use the default route point directory number.

Step 15 From the drop-down list, choose the appropriate numbering plan and then click **Next**.

Step 16 In the **Phone Services** window, enter the primary phone service name; otherwise, use the default Phone Service name.

Step 17 From the drop-down list, choose the primary Cisco Unified Communications Manager Assistant server or enter a server name or IP address.

Step 18 Enter the secondary phone service name; otherwise, use the default phone service name.

Step 19 From the drop-down list, choose the secondary Cisco Unified Communications Manager Assistant server or enter a server name or IP address and then click **Next**.
The **Confirmation** window is displayed. It provides all the information that you chose. If the information is not correct, you can cancel the configuration process or return to the previous configuration windows.

Step 20 Click **Finish**.
Upon completion, a final status window is displayed.

Any errors that the configuration wizard generates is sent to a trace file. Access this file by using the following CLI command: **`file get activelog tomcat/logs/ccmadmin/log4j`**

What to do next

The Cisco Unified CM Assistant Configuration Wizard only creates the Cisco IP Manager Assistant service parameters. You must enter the remaining service parameters manually. For service parameter information, see Manager Assistant Service Parameters for Proxy Line, on page 164.

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.

Setting	Description
Cisco IP Manager Assistant (Active) Parameters	
CTIManager (Primary) IP Address	This parameter specifies the IP address of the primary CTIManager that this Cisco IPMA process calls. No default value.
CTIManager (Backup) IP Address	This parameter specifies the IP address of the backup CTIManager that this Cisco IPMA process calls when primary CTIManager is down. No default value.
Route Point Device Name for Proxy Mode	This parameter specifies the device name of the CTI route point that this Cisco IPMA server all calls to managers' primary lines for intelligent call routing. Cisco recommends that you use same CTI route point device for all servers running the IP must configure the CTI route point device name if any manager or assistant will be configu mode.
CAPF Profile Instance Id for Secure Connection to CTIManager	This service parameter specifies the Instance ID of the Application CAPF Profile for the **IPMASecureSysUser** that this Manager Assistant will use to open a secure connection t Configure this parameter if **CTIManager Connection Security Flag** is enabled.
Clusterwide Parameters (Parameters that apply to all servers) **Important** Click **Advanced** to view the hidden parameters.	
Cisco IPMA Server (Primary) IP Address	This parameter specifies the IP address of the primary Cisco IPMA server. No default value.
Cisco IPMA Server (Backup) IP Address	This parameter specifies the IP address of the backup Cisco IPMA server. The backup se IPMA service when the primary IPMA server fails. No default value.

Setting	Description
Cisco IPMA Server Port	This parameter specifies the TCP/IP port on the Cisco IPMA servers to which the IPMA will open socket connections. You may change the parameter if a port conflict exists. Default value: 2912
Cisco IPMA Assistant Console Heartbeat Interval	This parameter specifies the interval, in seconds, at which the Cisco IPMA server sends (commonly referred to as heartbeat) to the IPMA Assistant Consoles. The IPMA Assist failover when they fail to receive heartbeat from the server before the time that is specif expires. Default value: 30 seconds
Cisco IPMA Assistant Console Request Timeout	This parameter specifies the time, in seconds, that the IPMA Assistant Consoles wait t from the Cisco IPMA server. Default value: 30 seconds
Cisco IPMA RNA Forward Calls	This parameter determines whether Cisco IPMA Ring No Answer (RNA) forwarding values are **True** (Cisco IPMA forwards unanswered calls to next available assistant) or does not forward calls). This parameter works in conjunction with the **Cisco IPMA RNA Timeout** parameter after the time that is specified in the **Cisco IPMA RNA Timeout** parameter. If a voic specified for the line, unanswered calls that cannot be forwarded to an assistant are ser this timer expires. Default value: False
Alpha Numeric UserID	This parameter determines whether Cisco IPMA Assistant Phone uses an alphanumeric user ID. Default value: True
Cisco IPMA RNA Timeout	This parameter specifies the time, in seconds, that the Cisco IPMA server waits befor unanswered call to the next available assistant. This parameter works in conjunction w **RNA Forward Calls** parameter; forwarding occurs only if the **Cisco IPMA RNA Forw** is set to **True**. Default value: 10 seconds
CTIManager Connection Security Flag	This parameter determines whether security for the Cisco IP Manager Assistant servic connection is enabled. If it is enabled, Cisco IPMA opens a secure connection to CTI CAPF profile that is configured for the instance ID (as specified in the **CAPF Profile Secure Connection to CTIManager** service parameter) for the application user **IPM** Default value: Non Secure To enable security, you must select an instance ID in the **CAPF Profile Instance ID for to CTIManager** service parameter.
Redirect call to Manager upon failure to reach Assistant	This parameter determines whether the Cisco Unified IP Manager Assistant applicati back to the intended manager if the call fails to reach the selected proxy assistant. Default value: False

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 tha	
Enable Multiple Active Mode	This parameter determines whether multiple instances of the Cisco IP Manager Assistant run for scalability. If it is enabled, Cisco IPMA can run on the other nodes as configured Pool 3 parameters. To enable multiple active mode, you must enter the IP addresses of the nodes on which y the additional instances of Cisco IPMA. Configure the Cisco IP Manager Assistant servic those nodes. Default value: False
Pool 2: Cisco IPMA Server (Primary) IP Address	If multiple active mode is enabled, this parameter specifies the IP address of the primary Ci of the second instance of Cisco IPMA. Configure the Cisco IP Manager Assistant service parameters on this node.
Pool 2: Cisco IPMA Server (Backup) IP Address	If multiple active mode is enabled, this parameter specifies the IP address of the backup Ci of the second instance of Cisco IPMA. The backup server provides IPMA service when th server fails. Configure the Cisco IP Manager Assistant service parameters on this node.
Pool 3: Cisco IPMA Server (Primary) IP Address	If multiple active mode is enabled, this parameter specifies the IP address of the primary Ci of the third instance of Cisco IPMA. Configure the Cisco IP Manager Assistant service parameters on this node.
Pool 3: Cisco IPMA Server (Backup) IP Address	If multiple active mode is enabled, this parameter specifies the IP address of the primary Ci of the third instance of Cisco IPMA. The backup server provides IPMA service when the server fails. Configure the Cisco IP Manager Assistant service parameters on this node.
Clusterwide Parameters (Softkey Templates) **Important** Configure these parameters if you want to use the Manager Assistant automatic configuration for managers and assistan	
Assistant Softkey Template	This parameter specifies the assistant softkey template that is assigned to assistant devices d Configuration. The value that is specified in this parameter is used when the **Automatic** check box is checked on the **Cisco IPMA Assistant Configuration** page.
Manager Softkey Template for Proxy Mode	This parameter specifies the manager softkey template for proxy mode that is assigned to during Automatic Configuration. This parameter applies only for managers that use prox
Clusterwide Parameters (IPMA Device Configuration Defaults for Proxy Mode)	
Manager Partition	This parameter defines the partition that is assigned to manager lines that IPMA handles on during Automatic Configuration. Make sure the partition you want to use has already bee Unified CM Administration. If the **Cisco IPMA Configuration Wizard** is run, it will po This parameter applies only for managers that use proxy mode.

Setting	Description
All User Partition	This parameter specifies the partition that is configured on all proxy lines and the inter devices, as well as the intercom line on manager devices, during Automatic Configur partition you want to use has already been added to Cisco Unified CM Administration **Configuration Wizard** is run, it will populate this value. This parameter applies only assistants that use proxy mode.
IPMA Calling Search Space	This parameter specifies the calling search space that is configured for manager lines that IPMA handles and the intercom line, as well as the assistant intercom line on ass Automatic Configuration. Make sure the calling search space you want to use has alr Cisco Unified CM Administration. If **Cisco IPMA Configuration Wizard** is run, it value. This parameter applies only for managers or assistants that use proxy mode.
Manager Calling Search Space	This parameter defines the manager calling search space that is configured on proxy l devices during Automatic Configuration. This calling search space must be a calling already exists in the system. If **Cisco IPMA Configuration Wizard** is run, it will pop parameter applies only for assistants that use proxy mode.
Cisco IPMA Primary Phone Service	This parameter defines the IP phone service to which manager/assistant devices will Automatic Configuration. If **Cisco IPMA Configuration Wizard** is run, it will popu parameter applies only for managers or assistants that use proxy mode.
Cisco IPMA Secondary Phone Service	This parameter defines the secondary IP phone service to which manager or assistant subscribed during Automatic Configuration. If **Cisco IPMA Configuration Wizard** i this value. This parameter applies only for managers or assistants that use proxy mod
Clusterwide Parameters (Proxy Directory Number Range for Proxy Mode)	
Starting Directory Number	This parameter specifies the starting directory number that is used as the starting num generation of proxy directory numbers during IPMA assistant configuration. After an a line number is used for an assistant, the next number will be generated for the next as This parameter applies only for assistants that use proxy mode.
Ending Directory Number	This parameter specifies the ending directory number for automatic generation of prox during IPMA assistant configuration. Configuration will stop at this number. This par for assistants that use proxy mode.
Clusterwide Parameters (Proxy Directory Number Range for Proxy Mode)	
Number of Characters to be Stripped from Manager DN	This parameter specifies the number of characters to be stripped from the manager dir in the process of generating the proxy DN. Generating a proxy DN may involve stripp digits and adding a prefix. Digits are stripped starting from the left. This parameter appli that use proxy mode.
Prefix for Manager DN	This parameter specifies the prefix to be added to a manager DN in the process of ge DN. Generating a proxy DN may involve some stripping of digits and adding a prefix applies only for assistants that use proxy mode.

Configure Manager And Assign Assistant For Proxy Line

For information about configuring a new user and associating a device to the user, see Administration Guide for Cisco Unified Communications Manager.

Note Make sure you configure manager information before you configure assistant information for an assistant.

Procedure

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

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

Step 3 From the **Related Links** drop-down list, choose **Manager Configuration** and click **Go**.

Tip
To view existing assistant configuration information, click the assistant name in the Associated Assistants list and click **View Details**. The **Cisco Unified CM Assistant - Assistant Configuration** window is displayed. To return to the manager configuration information, click the manager name in the Associated Managers list and click **View Details**.

The **Cisco Unified CM Assistant - Manager Configuration** window is displayed.

Step 4 From the **Device Name/Profile** drop-down list, choose the device name or device profile to associate a device name or device profile with a manager. For more information about Extension Mobility with Manager Assistant, see Manager Assistant Interactions, on page 188.

Note
If the manager telecommutes, click the **Mobile Manager** check box and optionally choose a device profile from the **Device Name/Profile** drop-down list. After you choose a device profile, the manager must log in to the phone by using extension mobility before accessing Manager Assistant.

Step 5 From the **Intercom Line** drop-down list, choose the intercom line appearance for the manager, if applicable.

Note
The chosen intercom line applies to the Manager Assistant and Unified Communications Manager intercom features.

Step 6 From the **Assistant Pool** drop-down list, choose the appropriate pool number (1 to 3).

Step 7 From the **Available Lines** selection box, choose a line that you want Manager Assistant to control and click the down arrow to make the line display in the Selected Lines selection box. Configure up to five Manager Assistant—controlled lines.

Tip
To remove a line from the Selected Lines selection box and from Manager Assistant control, click the up arrow.

Step 8 Check the **Automatic Configuration** check box to automatically configure the softkey template, subscribe to the Manager Assistant phone service, calling search space, and partition for Manager Assistant—Controlled selected lines and intercom line; and Auto Answer with Speakerphone for intercom line for the manager phone based on the Cisco IP Manager Assistant service parameters.

Note
Automatic Configuration for intercom applies only when using the Manager Assistant intercom feature for the Cisco Unified IP Phones 7940 and 7960.

Step 9 Click **Save**.
If you checked the **Automatic Configuration** check box and the service parameters are invalid, a message displays. Ensure that the service parameters are valid. Upon successful completion of the automatic configuration, the manager device resets. If you configured a device profile, the manager must log out and log in to the device for settings to take effect.

Configure Assistant Line Appearances for Proxy Line

A proxy line specifies a phone line that appears on the assistant Cisco Unified IP Phone. Manager Assistant uses proxy lines to manage calls that are intended for a manager. The administrators can manually configure a line on the assistant phone to serve as the proxy line, or you can enable the **Automatic Configuration** check box to generate a DN and to add the line to the assistant phone.

Note
1. Make sure that you configure manager information and assign an assistant to the manager before you configure assistant information for an assistant.
2. If you want to automatically configure proxy line on the assistant phone, configure the service parameters in **Proxy Directory Number Range** and **Proxy Directory Number Prefix** sections.

Procedure

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

Step 2 Click **Find**.

Step 3 Click on the user name to display user information for the chosen assistant
The **End User Configuration** window is displayed.

Step 4 From the **Related Links** drop-down list, choose **Assistant Configuration** and click **Go**.

Note
The system automatically sets the softkey template and intercom line on the basis of the Cisco IP Manager Assistant service parameter settings when the **Automatic Configuration** check box is checked. In addition, the system also sets Auto Answer with Speakerphone for intercom line.

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

Before you begin

- Review Manager Assistant Prerequisites, on page 161.

Procedure

	Command or Action	Purpose
Step 1	Configure Partitions for Manager Assistant Shared Line Support, on page 171	Configure a partition for lines that is used by Manager Assistant.
Step 2	Configure Calling Search Spaces for Manager Assistant Shared Line Support, on page 172	Configure calling search spaces for manager and assistant lines.
Step 3	Configure Cisco IP Manager Assistant Service Parameters, on page 173	Configure these parameters to use automatic configuration for managers and assistants.
Step 4	Configure intercom settings. • Configure an Intercom Partition, on page 174 • Configure an Intercom Calling Search Space, on page 313 • Configure an Intercom Directory Number, on page 314 • Configure an Intercom Translation Pattern, on page 314	
Step 5	Configure Multiple Manager Assistant Pool, on page 176	Configure multiple pools if you need to support a large number of managers and assistants. You can configure up to three active Cisco IP Manager Assistant servers, with each managing up to 2500 pairs of managers and assistants.
Step 6	Configure Secure TLS Connection to CTI • Configure IPMASecureSysUser Application User, on page 178 • Configure CAPF Profile, on page 178	Follow these procedures if your system is running in mixed mode.

	Command or Action	Purpose
	• Configure Cisco WebDialer Web Service , on page 180	
Step 7	Configure CTI Route Point, on page 180	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 181	
Step 9	Configure Phone Button Templates for Manager, Assistant, and Everyone, on page 184	
Step 10	Configure Manager and Assign Assistant for Shared Line Mode, on page 185	
Step 11	Configure Assistant Line Appearances for Shared Line, on page 186	
Step 12	Install Assistant Console Plugin, on page 187	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.	See Cisco Unified Communications Manager Assistant User Guide for Cisco Unified Communications Manager.

Configure Partitions for Manager Assistant Shared Line Support

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

Procedure

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

Step 2 Click **Add New** to create a new partition.

Step 3 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 4 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 5 To create multiple partitions, use one line for each partition entry.

Step 6 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 7 Select one of the following radio buttons to configure the **Time Zone**:

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

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 20: Partition Name Guidelines

Partition Name Length	Maximum Number of Partitions
2 characters	340
3 characters	256
4 characters	204
5 characters	172
...	...
10 characters	92
15 characters	64

Configure Calling Search Spaces for Manager Assistant Shared Line Support

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

You must create two calling search spaces: Generated_CSS_I_E and Generated_CSS_M_E.

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

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

Step 4 In the **Description** field, enter a description.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

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

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

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

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.

Procedure

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

Step 2 From the **Server** drop-down list, choose the server on which the Cisco IP Manager Assistant service is active.

Step 3 From the **Service** drop-down list, choose **Cisco IP Manager Assistant** service.
The **Service Parameter Configuration** window, which lists the parameters, is displayed.

Step 4 Configure the **Cisco IP Manager Assistant Parameters**, **Clusterwide Parameters (Parameters that apply to all servers)**, and **Clusterwide Parameters (Softkey Templates)**.

click ? for detailed descriptions.

Step 5 Click **Save**.

Configure Intercom Settings

Procedure

	Command or Action	Purpose
Step 1	Configure an Intercom Partition, on page 174	
Step 2	Configure an Intercom Calling Search Space, on page 175	
Step 3	Configure an Intercom Directory Number, on page 175	
Step 4	Configure an Intercom Translation Pattern, on page 176	

Configure an Intercom Partition

Procedure

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

The **Find and List Intercom Partitions** window appears.

Step 2 Click **Add New**.

An **Add New Intercom Partition** window appears.

Step 3 Under the **Intercom Partition Information** section, in the **Name** box, enter the name and description of the intercom partition that you want to add.

Note
To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (,) to separate the partition name and description on each line. If a description is not entered, Unified Communications Manager uses the partition name as the description.

Step 4 Click **Save**.

Step 5 Locate the partition that you want to configure.
Intercom Partition Configuration window is displayed

Step 6 Configure the fields in the Intercom Partition Configuration field area. See the online help for more information about the fields and their configuration options.

Step 7 Click **Save**.

The **Intercom Partition Configuration** window appears.

Step 8 Enter the appropriate settings. For detailed information about the Intercom Partition Configuration parameters, see online help.

Step 9 Click **Save**.

Step 10 Click **Apply Config**.

Configure an Intercom Calling Search Space

Procedure

Step 1 In the menu bar, choose **Call Routing** > **Intercom** > **Intercom Calling Search Space**.

Step 2 Click the **Add New**.

Step 3 Configure the fields in the Intercom Calling Search Space field area. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure an Intercom Directory Number

Procedure

Step 1 Choose **Call Routing** > **Intercom** > **Intercom Directory Number**.

The **Find and List Intercom Directory Numbers** window is displayed.

Step 2 To locate a specific intercom directory number, enter search criteria and click **Find.**
A list of intercom directory numbers that match the search criteria displayed.

Step 3 Perform one of the followings tasks:

a) To add an intercom directory number, click **Add New**.

b) To update an intercom directory number, click the intercom directory number to update.

The **Intercom Directory Number Configuration** window displayed.

Step 4 Configure the fields in the Intercom Directory Number Configuration field area. For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save**.

Step 6 Click **Apply Config**.

Step 7 Click **Reset Phone**.

Step 8 Restart devices.

During the restart, the system may drop calls on gateways.

Configure an Intercom Translation Pattern

Procedure

Step 1 Choose **Call Routing** > **Intercom** > **Intercom Translation Pattern**.

The **Find and List Intercom Translation Patterns** window appears.

Step 2 Perform one of the followings tasks:

a) To copy an existing intercom translation pattern, locate the partition to configure, click **Copy** eside the intercom translation pattern to copy.

b) To add a new intercom translation pattern, click the **Add New**.

Step 3 Configure the fields in the Intercom Translation Pattern Configuration field area. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Ensure that the intercom translation pattern that uses the selected partition, route filter, and numbering plan combination is unique. if you receive an error that indicates duplicate entries, check the route pattern or hunt pilot, translation pattern, directory number, call park number, call pickup number, or meet-me number configuration windows.

The **Intercom Translation Pattern Configuration** window displays the newly configured intercom translation pattern.

What to do next

Refer to the Manager Assistant Task Flow for Shared Lines, on page 170 to determine the next task to complete.

Configure Multiple Manager Assistant Pool

Procedure

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

Step 2 From the **Server** drop-down list, choose the server on which the Cisco IP Manager Assistant service is active.

Step 3 From the **Service** drop-down list, choose the **Cisco IP Manager Assistant** service.
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.

click ? for detailed descriptions.

Step 6 Click **Save**.

What to do next

Refer to the Manager Assistant Task Flow for Shared Lines, on page 170 to determine the next task to complete.

Configure Secure TLS Connection to CTI for Manager Assistant

Manager Assistant uses WDSecureSysUser application user credentials to establish a secure TLS connection to CTI to make calls.

To configure the WDSecureSysUser application user to establish a secure TLS connection, complete the following tasks.

Before you begin

- Install and configure the Cisco CTL Client.

  For more information about CTL Client, see Security Guide for Cisco Unified Communications Manager.
- Verify that the **Cluster Security Mode** in the **Enterprise Parameters Configuration** window is **1** (mixed mode). Operating the system in mixed mode impacts other security functions in your system. If your system is not currently running in mixed mode, do not switch to mixed mode until you understand these interactions. For more information, see Security Guide for Cisco Unified Communications Manager.
- Verify that the **Cluster SIPOAuth Mode** field in the **Enterprise Parameters Configuration** window is set to **Enabled**.
- Activate the Cisco Certificate Authority Proxy Function (CAPF) service on the first node.

Procedure

	Command or Action	Purpose
Step 1	Configure IPMASecureSysUser Application User, on page 178	Configure IPMASecureSysUser Application User.
Step 2	Configure CAPF Profile, on page 178	Configure Certificate Authority Proxy Function (CAPF) Profile for the IPMASecureSysUser Application User.
Step 3	Configure Cisco WebDialer Web Service , on page 180	Configure service parameters for the Cisco IP Manager Assistant service.

Configure IPMASecureSysUser Application User

Use this procedure to configure IPMASecureSysUser application user.

Procedure

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

Step 2 Click **Find**.

Step 3 From the **Find and List Application Users Application** window, choose **WDSecureSysUser**.

Step 4 Configure the fields in the **Application User Configuration** window and click **Save**.

Configure CAPF Profile

Certificate Authority Proxy Function (CAPF) is a component that performs tasks to issue and authenticate security certificates. When you create an application user CAPF profile, the profile uses the configuration details to open secure connections for the application.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **Application User CAPF Profile**.

Step 2 Perform one of the following tasks:

- Click **Add New** in the **Find** window, to add a new CAPF profile.
- Click **Copy** for that record in the **Copy** column, to copy an existing profile, and locate the appropriate profile.

To update an existing entry, locate and display the appropriate profile.

Step 3 Configure or update the relevant CAPF profile fields. See the Related Topics section information about the fields and their configuration options.

Step 4 Click **Save**.

Step 5 Repeat the procedure for each application and end user that you want to use security.

CAPF Profile Settings

Setting	Description
Application User	From the drop-down list, choose the application user for the CAPF operation. This setting displays configured application users. This setting does not appear in the **End User CAPF Profile** window.
End User ID	From the drop-down list, choose the end user for the CAPF operation. This setting displays configured end users. This setting does not appear in the **Application User CAPF Profile** window.

Setting	Description
Instance ID	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.
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 applicationPC. 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-to10-digit authentication string appears in the **Authentication String** field.
Key Size (bits)	From the drop-down list, choose the key size for the certificate. The default setting is 1024. The other option for key size is 512. Key generation, which is set at low priority, allows the application to function while the action occurs. Key generation may take up to 30 or more minutes.
Operation Completes by	This field, which supports all certificate operations, specifies the date and time by which you must complete the operation. The values that are displayed apply for the first node. Use this setting with the **CAPF Operation Expires in (days)** enterprise parameter, which specifies the default number of days in which the certificate operation must be completed. You can update this parameter at any time.
Certificate Operation Status	This field displays the progress of the certificate operation, such as pending, failed, or successful. You cannot change the information that is displayed in this field.

Configure Cisco WebDialer Web Service

Procedure

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

Step 2 From the **Server** drop-down list, choose the server on which the Cisco WebDialer Web service is active.

Step 3 From the **Service** drop-down list, choose the **Cisco WebDialer Web** service.
A list of parameters appears.

Step 4 Navigate to and update the CTIManager Connection Security Flag and CAPF Profile Instance ID for Secure Connection to CTIManager parameters.

To view parameter descriptions, click the parameter name link.

Note
CTIManager supports IPv4 and IPv6 addresses.

Step 5 Click **Save**.

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

What to do next

Refer to the Manager Assistant Task Flow for Shared Lines, on page 170 to determine the next task to complete.

Configure CTI Route Point

Procedure

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

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**.
`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**.

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 Cisco IP Phone Services Configuration Fields, on page 181 for more information about the fields and their configuration options.
The `Update successful` message is displayed.

Cisco IP Phone Services Configuration Fields

Field	Description
Service Information	
Service Name	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** 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. **Note** 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.
ASCII Service Name	Enter the name of the service to display if the phone cannot display Unicode.
Service Description	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 (") or single quotation marks (').

Field	Description
Service URL	Enter the URL of the server where the IP phone services application is located. Make sure that this server remains independent of the servers in your Unified Communications Manager cluster. Do not specify a Unified Communications Manager server or any server that is associated with Unified Communications Manager (such as a TFTP server or directory database publisher server). For the services to be available, the phones in the Unified Communications Manager cluster must have network connectivity to the server. For Cisco-signed Java MIDlets, enter the location where the JAD file can be downloaded; for example, a web server or the back-end application server to which the Java MIDlet communicates. For Cisco-provided default services, the service URL is displayed as `Application:Cisco/<name of service>` by default; for example, `Application:Cisco/CorporateDirectory`. If you modify the service URL for Cisco-provided default services, verify that you configured both for the **Service Provisioning** setting, which displays in the **Phone**, **Enterprise Parameter**, and **Common Phone Profile Configuration** windows. For example, you use a custom corporate directory, so you change `Application:Cisco/CorporateDirectory` to the external service URL for your custom directory; in this case, change the **Service Provisioning** value to **Both**.
Secure-Service URL	Enter the secure URL of the server where the Cisco Unified IP Phone services application is located. Make sure that this server remains independent of the servers in your Unified Communications Manager cluster. Do not specify a Unified Communications Manager server or any server that is associated with Unified Communications Manager (such as a TFTP server or publisher database server). For the services to be available, the phones in the Unified Communications Manager cluster must have network connectivity to the server. **Note** If you do not provide a Secure-Service URL, the device uses the nonsecure URL. If you provide both a secure URL and a nonsecure URL, the device chooses the appropriate URL, based on its capabilities.
Service Category	Choose a service application type (XML or Java MIDlet). If you choose Java MIDlet, when the phone receives the updated configuration file, the phone retrieves the Cisco-signed MIDlet application (JAD and JAR) from the specified Service URL and installs the application.
Service Type	Choose whether the service is provisioned to the Services, Directories, or Messages button or option on the phone; that is, if the phone has these buttons or options. To determine whether your phone supports these buttons or options, see the *Cisco Unified IP Phone Administration Guide* that supports your phone model.

Field	Description
Service Vendor	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.
Service Version	Enter the version number for the application. For XML applications, this field is optional and is informational only. For Cisco-signed Java MIDlets, consider the following information: • If you enter a version, the service version must exactly match the version that is defined in the JAD file. If you enter a version, the phone attempts to upgrade or downgrade the MIDlet if the version is different than what is installed on the phone. • If the field is blank, the version gets retrieved from the **Service URL**. Leaving the field blank ensures that the phone attempts to download the JAD file every time that the phone reregisters to Unified Communications Manager as well as every time that the Cisco-signed Java MIDlet is launched; this ensures that the phone always runs the latest version of the Cisco-signed Java MIDlet without you having to manually update the **Service Version** field. This field displays as blank for Cisco-provided default services. You can enter numbers and periods in this field (up to 16 ASCII characters).
Enable	Allows you to enable or disable the service without removing the configuration from Cisco Unified CM Administration (and without removing the service from the database). Uncheck the check box to remove the service from the phone configuration file and the phone.
Service Parameter Information	

Field	Description
Parameters	Lists the service parameters that apply to this IP phone service. Use the following buttons to configure service parameters for this pane: • **New Parameter**—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. • **Edit Parameter**—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. • **Delete Parameter**—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.

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

	Command or Action	Purpose
Step 1	Configure a Phone Button Template for Manager Assistant, on page 184	Perform this step to assign manage and assistant button features to line or speed dial keys.
Step 2	Associate a Manager Assistant Button Template with a Phone, on page 185	Perform this step to configure the manager and assistant button for a phone.

Configure a Phone Button Template for Manager Assistant

Use this procedure to configure a phone button template for the Manager Assistant feature.

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.

b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate a Manager Assistant Button Template with a Phone

Before you begin

Configure a Phone Button Template for Manager Assistant, on page 184

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure Manager and Assign Assistant for Shared Line Mode

Procedure

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

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

Step 3 From the **Related Links** drop-down list, choose **Manager Configuration** and click **Go**.

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 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 Unified Communications Manager intercom features.

Step 8 From the **Assistant Pool** drop-down list, choose the appropriate pool number (1 to 3).

Step 9 Choose the name of the assistant from the Available Assistants selection box and move it to the Associated Assistants selection box by clicking the down arrow to assign an assistant to the manager.

You can go to the **Assistant Configuration** window by highlighting the assistant name and clicking the **View Details** link.

Step 10 Choose the appropriate line from the Available Lines list box and move it to the Selected Lines list box by clicking the down arrow to configure the Manager Assistant controlled lines.
Make sure that the controlled line is always the shared line DN.

Step 11 Click **Save**.
If you checked the **Automatic Configuration** check box and the service parameters are invalid, a message is displayed. Ensure that the service parameters are valid. After successful completion of the automatic configuration, the manager device resets. If you configured a device profile, the manager must log out and log in to the device for the changes to take effect.

Configure Assistant Line Appearances for Shared Line

Administrators can set up one or more lines with a shared line appearance. The Unified Communications Manager system considers a directory number to be a shared line if it appears on more than one device in the same partition.

Procedure

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

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

Step 3 Click on the username to display user information for the chosen assistant.
The **End User Configuration** window is displayed.

Step 4 From the **Related Links** drop-down list, choose **Assistant Configuration** and click **Go**.
The **Assistant Configuration** window is displayed. The system automatically sets the softkey template and intercom line on the basis of the Cisco IP Manager Assistant service parameter settings when you check the **Automatic Configuration** check box. In addition, the system also sets Auto Answer with Speakerphone for intercom line.

Step 5 From the **Device Name** drop-down list, choose the device name to associate with the assistant.

Step 6 From the **Intercom Line** drop-down list, choose the incoming intercom line appearance for the assistant.

Step 7 From the **Primary Line** drop-down list, choose the primary line for the assistant.

a) To view existing manager configuration information, highlight the manager name in the **Associated Managers** list and click **View Details**.
The **Manager Configuration** window is displayed.

b) To return to the **Assistant Configuration** window, highlight the assistant name and click **View Details** link in the **Manager Configuration** window.

In the **Associated Manager** selection list box, the name of the previously configured manager is displayed.

Step 8 To associate the manager line to the assistant line, perform the following steps from the Manager Association to Assistant Line selection box:

a) From the **Available Lines** drop-down list, choose the assistant line that will be associated with the manager line.

b) From the **Manager Names** drop-down list, choose the preconfigured manager name for whom this proxy line will apply.

c) From the **Manager Lines** drop-down list, choose the manager line for which this proxy line will apply.

Step 9 Click **Save**.
The update takes effect immediately. If you chose **Automatic Configuration**, the assistant device automatically resets.

Install Assistant Console Plugin

Procedure

Step 1 From Cisco Unified CM Administration, choose **Application** > **Plugins**.
The **Find and List Plugins** window is displayed.

Step 2 Click **Find**.
A list of installable application plug-ins is displayed.

Step 3 Click on the **Download** link for Cisco Unified CM Assistant Console and save the executable to a location.

Step 4 Run the executable file.

Note
If you install the application on a Windows Vista PC, a security window may be displayed. Allow the installation to continue.

The **Cisco Unified CallManager Assistant Console** installation wizard is displayed.

Step 5 In the **Introduction** window, click **Next**.

Step 6 In the **License Agreement** window, click **Next**.

Step 7 Choose a location where you want the application to install and click **Next**.

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 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

Feature	Interaction
Bulk Administration Tool	You can use the Bulk Administration Tool to add many users (managers and assistants) at once instead of adding users individually. The Bulk Administration Tool templates that the **Cisco Unified CM Assistant Configuration Wizard** creates for Cisco Unified IP Phones support only the Unified Communications Manager intercom lines. For more information, see the Bulk Administration Guide for Cisco Unified Communications Manager.
Calling Party Normalization	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.
Extension Mobility	You can simultaneously use Manager Assistant with the Cisco Extension Mobility feature. When you log in to the Cisco Unified IP Phone using Extension Mobility, the Cisco IP Manager Assistant service is automatically enabled on that phone. You can then access the Manager Assistant features. For more information about Cisco Extension Mobility, see Extension Mobility Overview, on page 403.
Internet Protocol Version 6 (IPv6)	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.

Feature	Interaction
Reporting tools	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*.
CDR Analysis and Reporting	Manager Assistant supports call-completion statistics and inventory reporting for managers and assistants. The CAR tool supports call-completion statistics. Cisco Unified Serviceability supports inventory reporting. For more information, see the following guides: • *Cisco Unified Serviceability Administration Guide* • Call Reporting and Billing Administration Guide for Cisco Unified Communications Manager
Multilevel Precedence and Preemption (MLPP)	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.

Feature	Interaction
Intercom	Manager Assistant supports the following two types of intercom: • 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. • 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.
Message Waiting Indicator	The Message Waiting Indicator feature interacts with proxy line support only. 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.
Time-of-Day Routing	The Time-of-Day feature interacts with proxy line support only. 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. For more information about Time-of-Day Routing, see the System Configuration Guide for Cisco Unified Communications Manager.

Manager Assistant Restrictions

Feature	Restriction
Assistant Console Application	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.
Call Management features	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).

Feature	Restriction
Cisco IP Phones	Manager Assistant supports SIP on Cisco Unified IP Phones 7900 Series, except for Cisco Unified IP Phones 7940 and 7960. Manager Assistant supports up to 3500 managers and 3500 assistants by configuring multiple Cisco IP Manager Assistant servers (pools). When you enable multiple pools, the manager and all configured assistants for that manager should belong to the same pool. Cisco Unified IP Phones 7960 and 7940 support only the Unified Communications Manager Assistant Intercom lines feature. Cisco Unified IP Phones 7900 (except 7940 and 7960) support only the Unified Communications Manager Intercom feature. One manager can have up to ten assigned assistants and one assistant can support up to 33 managers (if each manager has one Unified Communications Manager–controlled line). Only one assistant at a time can assist a manager. Manager Assistant supports up to 3500 managers and 3500 assistants per Unified Communications Manager cluster.
Intercom	After an upgrade, Manager Assistant users that use the incoming intercom line do not get upgraded automatically to the Unified Communications Manager Intercom feature. The system does not support calls between the Unified Communications Manager Intercom feature and regular lines (which may be configured as Manager Assistant Intercom lines).
Single Sign-On	Manager Assistant is not supported in the Single Sign-On environment.
Speed Dial	Cisco Unified IP Phones 7940, 7942, and 7945 support only two lines or speed-dial buttons.

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.

Tool Description	Location
Cisco Unified CM Assistant server trace files	The trace files reside on the server that runs the Cisco IP Manager Assistant service. You can download these files from the server using one of the following methods: • Use the CLI command **file get activelog tomcat/logs/ipma/log4j**. • Use the trace collection features in the Cisco Unified Real-Time Monitoring Tool (RTMT). For more information, see the *Cisco Unified Real-Time Monitoring Tool Administration Guide*. You can enable debug tracing by choosing **Cisco Unified Serviceability** > **Trace** > **Configuration**.
Cisco IPMA client trace files	**$INSTALL_DIR\logs\ACLog*.txt** on the client desktop, in the same location where the Unified CM Assistant assistant console resides. To enable debug tracing, go to the **Settings** dialog box in the Assistant Console. In the **Advanced** panel, check the **Enable Trace** check box. **Note** This check box enables only debug tracing. Error tracing always remains On.
Cisco IPMA client install trace files	**$INSTALL_DIR\InstallLog.txt** on the client desktop, in the same location where the Assistant Console resides.
Cisco IPMA Client AutoUpdater trace files	**$INSTALL_DIR\UpdatedLog.txt** on the client desktop, in the same location where the Unified CM Assistant Console resides.
Install directory	By default—`C:\Program Files\Cisco\Unified Communications Manager Assistant Console\`

Calling Party Gets Reorder Tone

Problem

Calling party gets a reorder tone or a message:

```
This call cannot be completed as dialed.
```

Possible Cause

The calling search space of the calling line may not be configured correctly.

Solution

Check the calling search space of the line. For more information about configuration, see the System Configuration Guide for Cisco Unified Communications Manager.

You can also use the Cisco Dialed Number Analyzer service to check for flaws in the calling search space. For more information, see the *Cisco Unified Communications Manager Dialed Number Analyzer Guide*.

Calls Do Not Get Routed When Filtering Is On or Off

Problem

Calls are not routed properly.

Possible Cause 1

Cisco CTI Manager service may have stopped.

Solution 1

Restart the Cisco CTI Manager and Cisco IP Manager Assistant services from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Possible Cause 2

The Unified Communications Manager Assistant route point was not configured properly.

Solution 2

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

Possible Cause 3

The status window on the manager phone displays the message `Filtering Down`. This message can indicate that Unified Communications Manager Assistant CTI route point may be deleted or may not be in service.

Solution 3

Use the following procedure to configure the CTI route point and restart the Cisco IP Manager Assistant service:

1. From Cisco Unified CM Administration, choose **Device** > **CTI Route Point**.
2. Find the route point, or add a new route point. For more information about configuration, see the System Configuration Guide for Cisco Unified Communications Manager.
3. Restart the Cisco CTI Manager and Cisco IP Manager Assistant services from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Cisco IP Manager Assistant Service Unreachable

Problem

After you open the Assistant Console, the following message is displayed:

```
Cisco IPMA Service Unreachable
```

Possible Cause 1

Cisco IP Manager Assistant service may have stopped.

Solution 1

Restart the Unified Communications Manager Assistant from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Possible Cause 2

The server address for the primary and secondary Unified Communications Manager Assistant servers may be configured as DNS names, but the DNS names are not configured in the DNS server.

Solution 2

Use the following procedure to replace the DNS name.

1. From Cisco Unified CM Administration, choose **System** > **Server**.
2. Replace the DNS name of the server with the corresponding IP address.
3. Restart the Unified Communications Manager Assistant from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Possible Cause 3

The Cisco CTI Manager service may have stopped.

Solution 3

Restart the Unified Communications Manager Assistant from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Possible Cause 4

The Unified Communications Manager Assistant service might be configured to open a CTI connection in secure mode, but the security configuration may not be complete.

If this scenario occurs, the following message is displayed in the alarm viewer or in the Unified Communications Manager Assistant service logs:

```
IPMA Service cannot initialize - Could not get Provider.
```

Solution 4

Check the security configuration in the service parameters of Cisco IP Manager Assistant service.

Restart the Unified Communications Manager Assistant from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Cannot Initialize Cisco IP Manager Assistant Service

Problem

The Cisco IP Manager Assistant service cannot open a connection to CTI Manager, and the following message is displayed:

```
IPMA Service cannot initialize - Could not get Provider
```

Possible Cause

The Cisco IP Manager Assistant service cannot open a connection to CTIManager. You can see the message in the alarm viewer or in the Unified CM Assistant service logs.

Solution

Restart the Cisco CTI Manager and Cisco IP Manager Assistant services from **Cisco Unified Serviceability** > **Tools** > **Control Center—Feature Services**.

Assistant Console Installation from Web Fails

Problem

Assistant Console installation from the web fails. The following message is displayed:

```
Exception: java.lang.ClassNotFoundException: InstallerApplet.class
```

Possible Cause

Using the Sun Java plug-in virtual machine instead of the Microsoft JVM with the standard Unified Communications Manager Assistant Console install causes failures.

Solution

The administrator directs the user to the following URL, which is a JSP page that supports the Sun Java plug-in:

```
https://<servername>:8443/ma/Install/IPMAConsoleInstallJar.jsp
```

HTTP Status 503—This Application Is Not Currently Available

Problem

http://<server-name>:8443/ma/Install/IPMAConsoleInstall.jsp displays the following error message:

```
HTTP Status 503—This application is not currently available
```

Possible Cause

Cisco IP Manager Assistant service has not been activated or is not running.

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 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 Unified Communications Manager Assistant, the service still runs. The display on the manager IP phone disappears. Calls do not get routed, although filtering is On. To verify that the manager is logged out, view the application log using the Cisco Unified Real-Time Monitoring Tool. Look for a warning from the Cisco Java Applications that indicates that the Cisco IP Manager Assistant service logged out.

Possible Cause

The manager pressed the softkeys more than four times per second (maximum limit allowed).

Solution

The Unified Communications Manager administrator must update the manager configuration. Perform the following procedure to correct the problem:

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

   The **Find and List Users** window is displayed.
2. Enter the manager name in the search field and click **Find**.
3. From the search results list, choose the manager that you want to update.

   The **End User Configuration** window is displayed.
4. From the **Related Links** drop-down list, choose **Cisco IPMA Manager** and click **Go**.
5. Make the necessary changes to the manager configuration and click **Update**.

Manager Cannot Intercept Calls That Are Ringing on the Assistant Proxy Line

Problem

The manager cannot intercept the calls that are ringing on the assistant proxy line.

Possible Cause

The calling search space of the proxy line is not configured properly.

Solution

Check the calling search space of the proxy line for the assistant phone. Perform the following procedure to correct the problem:

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

   The **Find and List Phones** search window is displayed.

2. Click the assistant phone.

   The **Phone Configuration** window is displayed.

3. Verify the calling search space configuration for the phone and for the directory number (line) and update as appropriate.

No Page Found Error

Problem

`http://<server-name>:8443/ma/Install/IPMAConsoleInstall.jsp` displays the following error message:

```
No Page Found Error
```

Possible Cause 1

Network problems.

Solution 1

Ensure that the client has connectivity to the server. Ping the server name that is specified in the URL and verify that it is reachable.

Possible Cause 2

Misspelled URL.

Solution 2

Because URLs are case sensitive, ensure that the URL matches exactly with the URL in the instructions.

System Error - Contact System Administrator

Problem

After you open the Assistant Console, the following message is displayed:

```
System Error - Contact System Administrator
```

Possible Cause 1

You may have upgraded the Unified Communications Manager. The system does not upgrade the Assistant Console automatically when you upgrade the Unified Communications Manager.

Solution 1

Uninstall the console by choosing `Start > Programs > Cisco Unified Communications Manager Assistant > Uninstall Assistant Console` and reinstall the console from URL **https://<server-name>:8443/ma/Install/IPMAConsoleInstall.jsp**.

Possible Cause 2

The user is not configured correctly in the database.

Solution 2

Ensure that the user ID and the password are run as a Unified Communications Manager user through Cisco Unified CM Administration.

Possible Cause 3

When you deleted a manager from an assistant, Cisco Unified CM Administration left a blank line for the assistant.

Solution 3

From the **Assistant Configuration** window, reassign the proxy lines.

Unable to Call Manager When Cisco IP Manager Assistant Service is Down

Problem

Calls do not get routed properly to managers when Cisco IP Manager Assistant service goes down.

Possible Cause

The Unified Communications Manager Assistant CTI route point does not have Call Forward No Answer enabled.

Solution

Perform the following procedure to properly configure the Unified Communications Manager Assistant route point.

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

   The **Find and List CTI Route Point** search window is displayed.

2. Click **Find**.

   A list of configured CTI route points is displayed.

3. Choose the Unified Communications Manager Assistant CTI route point that you want to update.

4. In the **CTI Route Point Configuration** window, click the line to update from the **Association** area.

5. In the **Call Forward and Pickup Settings** section, check the **Forward No Answer Internal** and the **Forward No Answer External** check box and enter the CTI route point DN in the **Coverage/Destination** field (for example, CFNA as 1xxx for the route point DN 1xxx).

6. In the **Calling Search Space** drop-down list, choose **CSS-M-E** (or appropriate calling search space).
7. Click **Update**.

User Authentication Fails

Problem

User authentication fails when you sign in using the login window from the Assistant Console.

Possible Cause

The following probable causes can apply:

- Incorrect management of the user in the database
- Incorrect management of the user as an assistant or a manager

Solution

Ensure that the user ID and the password are ran as a Unified Communications Manager user through Cisco Unified CM Administration.

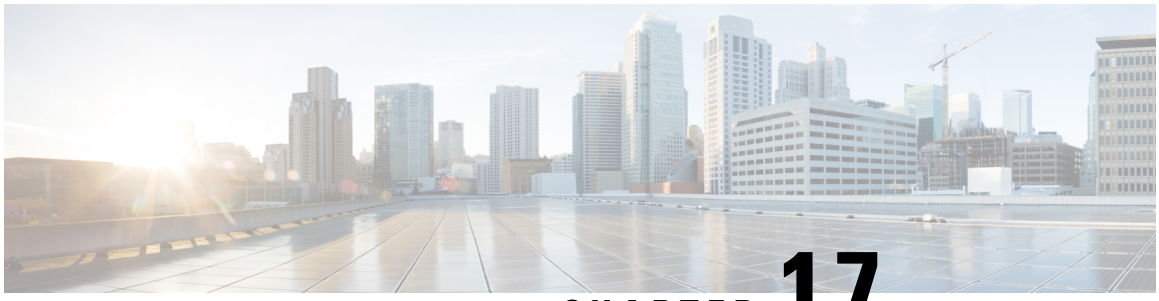
You must run the user as an assistant or a manager by associating the Unified Communications Manager Assistant user information, which you access through Cisco Unified CM Administration under **User Management** > **End User**.



PART VII

Voice Messaging Features

- Audible Message Waiting Indicator , on page 203
- Immediate Divert , on page 207



CHAPTER 17

Audible Message Waiting Indicator

- Audible Message Waiting Indicator Overview, on page 203
- Audible Message Waiting Indicator Prerequisites, on page 203
- Audible Message Waiting Indicator Configuration Task Flow, on page 203
- Audible Message Waiting Indicator Troubleshooting, on page 205

Audible Message Waiting Indicator Overview

You can configure Audible Message Waiting Indicator (AMWI) to play a stutter dial tone on the Cisco Unified IP Phone to notify users of new voice messages. Users hear a stutter dial tone whenever the phone goes off hook on a line on which a voice message was left.

You can configure AMWI for all the phones in a cluster or for only certain directory numbers. The directory-number-level configuration takes precedence over the cluster-wide configuration.

Audible Message Waiting Indicator Prerequisites

You can configure AMWI only on Cisco Unified IP Phone that are running phone firmware Release 8.3(1) or later.

Audible Message Waiting Indicator Configuration Task Flow

Before you begin

- Review Audible Message Waiting Indicator Prerequisites, on page 203.

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the Audible Message Waiting Indicator feature.

	Command or Action	Purpose
Step 2	Configure Audible Message Waiting Indicator Service Parameters, on page 204	Configure AMWI default setting for all phones in a cluster.
Step 3	Configure Audible Message Waiting Indicator for a Directory Number, on page 204	Configure AMWI for a directory number that is associated to a device.
Step 4	Configure Audible Message Waiting Indicator for a SIP Profile, on page 205	Configure AMWI for SIP profiles. Perform this procedure to configure AMWI for SIP phones.

Configure Audible Message Waiting Indicator Service Parameters

This procedure describes how to configure AMWI default setting for all the phones in a cluster.

Before you begin

Generate a Phone Feature List, on page 5

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 In the **Clusterwide Parameters (Feature - General)** section, choose the **Audible Message Waiting Indication Policy** service parameter. This parameter determines whether the Audible Message Waiting Indicator is turned on of off for all the devices in the cluster.

Step 5 Click **Save**.

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

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

Step 2 In the **Association** section, click **Add a new DN**.
The **Directory Number Configuration** window appears.

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.

Step 4 Configure the remaining fields in the **Directory Number Configuration** window. See the online help for more information about the fields and their configuration options.

Step 5 Click **Save**.

Configure Audible Message Waiting Indicator for a SIP Profile

Follow these steps to configure Audible Message Waiting Indicator (AMWI) for a SIP profile.



Note The AMWI setting on an individual SIP profile overrides the clusterwide setting.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.
The **Find and List SIP Profiles** window appears.

Step 2 Enter the search criteria to use and click **Find**.
The window displays a list of SIP profiles that match the search criteria.

Step 3 Click the SIP profile that you want to update.
The **SIP Profile Configuration** window appears.

Step 4 Check the **Stutter Message Waiting** check box to activate stutter dial tone when the phone is off hook and a message is waiting.

Step 5 Click **Save**.

Step 6 Click **Apply Config**.

Audible Message Waiting Indicator Troubleshooting

Audible Message Waiting Indicator Is Not Heard on the Phone

Problem Phone does not play stutter dial tone to notify the user of new voice messages.

If the user uses an SCCP phone, check the following:

- Ensure that the phone firmware release is 8.3(1) or later.
- Check the AMWI setting for the phone and the line on which the user went off hook.

- Verify that the Cisco CallManager service is running on the server.
- Check the sniffer trace between the phone and Unified Communications Manager. Make sure that the phone receives the StartTone message with tone type equal to 42.

If the user uses a SIP phone, check the following:

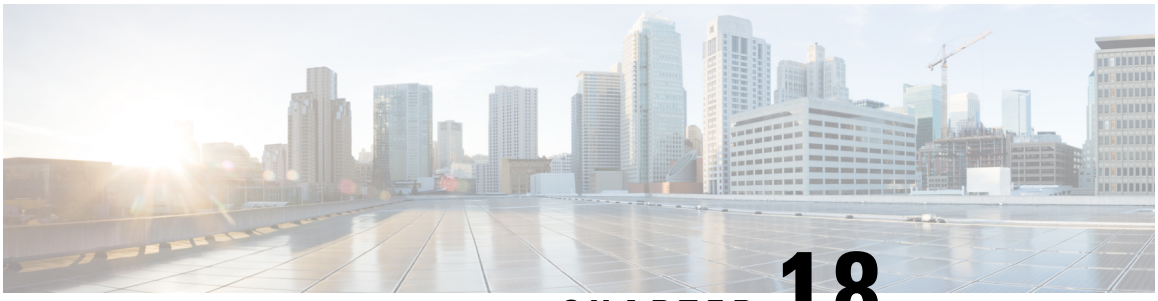
- Ensure that the phone firmware release is 8.3(1) or later.
- Check the line (directory number) configuration. The phone must display the settings such as line1_msgWaitingAMWI : 1, line2_msgWaitingAMWI : 0.
- Ensure that the **Stutter Message Waiting** check box is checked in the **SIP Profile Configuration** window in Cisco Unified CM Administration.

Localized AMWI Tone Is Not Played in a Specific Locale

Problem The phone that is configured in a non-English locale does not play the localized tone.

Solution Check the following:

- From Cisco Unified CM Administration, verify the User Locale in the **Device Profile Configuration** window (**Device** > **Device Settings** > **Device Profile**).
- Make sure that the user resets the phone after changing the locale.
- Check `user/local/cm/tftp /<locale name> directory` and verify that the AMWI tone is defined in the localized g3-tones.xml file.



CHAPTER 18

Immediate Divert

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- Immediate Divert Prerequisites, on page 208
- Immediate Divert Configuration Task Flow, on page 208
- Immediate Divert Interactions, on page 213
- Immediate Divert Restrictions, on page 214
- Immediate Divert Troubleshooting, on page 216

Immediate Divert Overview

The Immediate Divert feature is a Unified Communications Manager supplementary service that allows you to immediately divert a call to a voicemail system. When Immediate Divert diverts a call, the line becomes available to make or receive new calls. Access the Immediate Divert feature by using the iDivert or Divert softkey on the IP phone.

Immediate Divert provides the following functions:

- Diverts a call to a voicemail system in the following manner:
  - Legacy iDivert diverts the call to the voice mailbox of the party that invokes the iDivert feature.
  - Enhanced iDivert diverts the call to either the voice mailbox of the party that invokes the iDivert feature or to the voice mailbox of the original called party.
- Diverts inbound calls that are in the Call Offering, Call on Hold, or Call Active states.
- Diverts outbound calls in the Call Active or Call on Hold states.

Note Although the Immediate Divert feature is not available to CTI applications, a CTI redirect operation exists that performs the same function as Immediate Divert. Application developers can use the CTI redirect operation to accomplish Immediate Divert.

Immediate Divert Prerequisites

- You must configure the voicemail profiles and hunt pilots.

  For information on how to configure voicemail profiles and hunt pilots, see System Configuration Guide for Cisco Unified Communications Manager

- The following devices support Immediate Divert:
  - Voice-messaging systems such as Cisco Unity Connection that use the Skinny Client Control Protocol (SCCP).
  - QSIG devices (QSIG-enabled H.323 devices, MGCP PRI QSIG T1 gateways, and MGCP PRI QSIG E1 gateways), depending on the setting of the Use Legacy Immediate Divert and Allow QSIG During iDivert clusterwide service parameters.
  - The following table lists the phones that use the Divert or iDivert softkey.

Table 21: Cisco Unified IP Phones That Use Immediate Divert Softkeys

Cisco Unified IP Phone Model	Divert Softkey	iDivert Softkey	What to configure in softkey template
Cisco Unified IP Phone 6900 Series (except 6901 and 6911)	X		iDivert
Cisco Unified IP Phone 7900 Series		X	iDivert
Cisco Unified IP Phone 8900 Series	X		Configured by default
Cisco Unified IP Phone 9900 Series	X		Configured by default

Note Cisco Unified IP Phones 8900 and 9900 series have the Divert softkey assigned by default.

Immediate Divert Configuration Task Flow

Before you begin

- Review Immediate Divert Prerequisites, on page 208.

Procedure

	Command or Action	Purpose
Step 1	Configure Immediate Divert Service Parameters, on page 209	Configure the service parameters to enable Immediate Divert across various devices and applications.
Step 2	Configure a Softkey Template for Immediate Divert, on page 210	Create and configure a softkey template and add the iDivert softkey to that template.
Step 3	To Associate a Softkey Template with a Common Device Configuration, on page 211, complete the following subtasks: • Add a Softkey Template to the Common Device Configuration, on page 212 • Associate a Common Device Configuration with a Phone, on page 212	**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 213	**Optional**. Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in conjunction with the Common Device Configuration. Use this procedure in conjunction with the Common Device Configuration if you need assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Configure Immediate Divert Service Parameters

Procedure

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

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

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

Step 4 Configure the relevant service parameters and click **Save**.

Table 22: Service Parameter Fields for Immediate Divert

Field	Description
Call Park Display Timer	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.

Field	Description
Use Legacy Immediate Divert	Select one of the following options from the drop-down list: • **True**—The user that invokes the iDivert feature can divert an incoming call only to his own voice mailbox. This is the default setting. • **False**—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.
Allow QSIG During iDivert	Select one of the following options from the drop-down list: • **True**—Immediate Divert diverts calls to voicemail systems that can be reached over QSIG, SIP, and QSIG-enabled H.323 devices. • **False**—Immediate Divert does not support access to voicemail systems over QSIG or SIP trunks. This is the default setting.
Immediate Divert User Response Timer	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.

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**.

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

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 213

Procedure

	Command or Action	Purpose
Step 1	Add a Softkey Template to the Common Device Configuration, on page 212	
Step 2	Associate a Common Device Configuration with a Phone, on page 212	

Add a Softkey Template to the Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey Template with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Before you begin

Configure a Softkey Template for Immediate Divert, on page 210

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

Immediate Divert Interactions

Feature	Interaction
Multilevel Precedence and Preemption (MLPP)	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.
Call Forward	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).

Feature	Interaction
Call Detail Records (CDR)	Immediate Divert uses the immediate divert code number in the **Onbehalf** of fields (for example, joinOnbehalfOf and lastRedirectRediectOnBehalfOf) in CDR.
Call Park and Directed Call Park	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.
Conference	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.
Hunt List	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. **Note** For Jabber in desk phone mode, iDivert feature redirection to VM is done through CTI application where 'Use Legacy Immediate Divert' parameter will not take effect and HP number will be sent as diversion info to Voice mail servers.
Auto Call Pickup	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.

Immediate Divert Restrictions

Restriction	Description
Voice Mail Profile	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 **Make this the default Voice Mail Profile for the System** check box unchecked. Ensure the default Voice Mail Profile setting is always set to No Voice Mail.
Call Forward All (CFA) and Call Forward Busy (CFB)	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).

Restriction	Description
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.
Forward No Answer Timeout	A race condition in connection with the Forward No Answer Timeout exists when you press the iDivert softkey. For example, if a manager presses the iDivert softkey immediately after the Forward No Answer timeout, call forward forwards the call to a preconfigured directory number. However, if the manager presses the iDivert softkey before the Forward No Answer timeout, immediate divert diverts the call to the voice-messaging mailbox of the manager.
Calling Parties and Called Parties	The calling parties and called parties can divert the call to their voice mailboxes if both simultaneously press the iDivert softkey.
Conference Types	When one participant in a conference presses the iDivert softkey, all remaining participants receive an outgoing greeting of the participant who pressed iDivert. Conference types include Meet-Me, Ad Hoc, cBarge, and Join.
Split or Join Operation	If the last action on a call was Auto Pickup, Call Transfer, Call Park, Call Park Reversion, Conference, Meet-Me Conference, or any application that performs a split or join operation, enhanced iDivert does not present a screen to a called party to choose the voice mailbox. Instead, enhanced iDivert immediately diverts the call to the voice mailbox that is associated with the called party.

Immediate Divert Troubleshooting

Key is not active

The phone displays this message when the user presses iDivert:

```
Key is not active
```

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

Configure a voice-messaging pilot in the user voice-messaging profile.

Temporary Failure

The phone displays this message when the user presses iDivert:

```
Temporary Failure
```

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

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

Busy

The phone displays this message when the user presses iDivert:

```
Busy
```

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

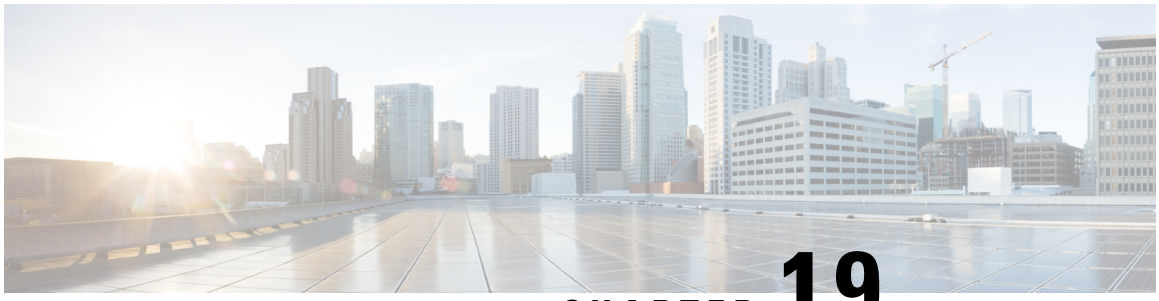
Configure more voice-messaging ports or try again.



PART VIII

Conferencing Features

- Ad Hoc Conferencing , on page 219
- Meet-Me Conferencing , on page 231
- Conference Now , on page 237



CHAPTER 19

Ad Hoc Conferencing

- Ad Hoc Conferencing Overview, on page 219
- Ad Hoc Conferencing Task Flow, on page 219
- Conference Interactions, on page 228
- Conference Restrictions, on page 228

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

	Command or Action	Purpose
Step 1	Configure Softkey Template for Conferencing, on page 220	Add the Conference List, Join, and Remove Last Conference Party softkeys to a softkey template.
Step 2	To Associate Softkey Template Common Device, on page 222, complete the following subtasks:	**Optional**. To make the softkey template available to phones, you must complete either this step or the following step. Follow this step

	Command or Action	Purpose
	• Add a Softkey Template to a Common Device Configuration, on page 222 • Associate a Common Device Configuration with a Phone, on page 223	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 223	**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 223	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 227	Enable Join Across Lines to create a conference.

Configure Softkey Template for Conferencing

Use this procedure to make the following conferencing softkeys available:

Softkey	Description	Call States
Conference List (**ConfList**)	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.	On Hook Connected
Join	Join up to 15 established calls (for a total of 16) to create a conference.	On Hold
Remove Last Conference Party (**Remove**)	The conference controller can invoke the conference list and remove any participant in the conference by using the **Remove** softkey.	On Hook Connected

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Complete one of the following procedures:

- Associate Softkey Template Common Device, on page 222
- Associate a Softkey Template with a Phone, on page 223

Associate Softkey Template Common Device

Optional. There are two ways to associate a softkey template with a phone:

- Add the softkey template to the **Phone Configuration**.
- Add the softkey template to the **Common Device Configuration**.

The procedures in this section describe how to associate the softkey template with a **Common Device Configuration**. Follow these procedures if your system uses a **Common Device Configuration** to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.

To use the alternative method, go to Associate a Softkey Template with a Phone, on page 223

Before you begin

Configure Softkey Template for Conferencing, on page 220

Procedure

	Command or Action	Purpose
Step 1	Add a Softkey Template to a Common Device Configuration, on page 222	Perform this step to add a conferencing softkey template to the Common Device Configuration.
Step 2	Associate a Common Device Configuration with a Phone, on page 223	Perform this step to link the conferencing softkey Common Device Configuration to a phone.

Add a Softkey Template to a Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey Template with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Configure 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. For parameter descriptions, see Ad Hoc Conferencing Service Parameters, on page 224.

Step 5 Click **Save.**

What to do next

Configure Join Across Lines, on page 227

Ad Hoc Conferencing Service Parameters

The following table lists the main service parameters for Ad Hoc conferencing. For additional conferencing service parameters, refer to the **Service Parameter Configuration** window's **Advanced** option. Conferencing service parameters appear under **Clusterwide Parameters (Feature - Conference)**.

Table 23: Ad Hoc Conference Service Parameters

Service Parameters	Description
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. (We recommend 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. Unified Communications Manager releases all resources that are assigned to the conference. **Note** Drop Ad Hoc Conference feature in an ILS deployment will not drop the parties when it set at **When No OnNet Parties Remain in the Conference** because the route patterns learned are classified as On Net. • **When Conference Controller Leaves**—The active conference terminates when the primary controller (conference creator) hangs up. Unified Communications Manager releases all resources that are assigned to the conference. **Note** We recommend 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
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.
Non-linear Ad Hoc Conference Linking Enabled	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).

Service Parameters	Description
Choose Encrypted Audio Conference Instead Of Video Conference	This parameter determines whether Unified Communications Manager chooses an encrypted audio conference bridge or an unencrypted video conference bridge for an Ad-Hoc conference call when the conference controller's Device Security Mode is set to either Authenticated or Encrypted and at least two conference participants are video-capable. Because encrypted video conference bridges are not supported in this release, Unified Communications Manager must choose between an encrypted audio conference bridge and an unencrypted video conference bridge. The default value is **True**.
Minimum Video Capable Participants To Allocate Video Conference	This parameter specifies the number of video-capable conference participants that must be present in an Ad Hoc conference to allocate a video conference bridge. If the number of video-capable participants is less than the number specified in this parameter, Unified Communications Manager allocates an audio conference bridge. If the number of video-capable participants is equal to, or greater than, the number specified in this parameter, Unified Communications Manager allocates a video conference bridge, when available, from the configured media resource group list (MRGL). Specifying a value of zero means that video conference bridges will always be allocated, even when none of the participants on the conference are video-capable. When a conference has been established using an audio bridge and then additional video-capable participants join the conference, the conference will remain on the audio bridge and will not convert to video. The default value is **2**.
Allocate Video Conference Bridge For Audio Only Conferences When The Video Conference Bridge Has Higher Priority	This parameter determines whether Unified Communications Manager chooses a video conference bridge, when available, for an Ad Hoc audio-only conference call when the video conference bridge has a higher priority than an audio conference bridge in the media resource group list (MRGL). If an audio conference bridge has higher priority than any video conference bridge in the MRGL, Unified Communications Manager ignores this parameter. This parameter proves useful in situations where the local conference bridge is a video bridge (and configured in the MRGL with the highest priority) and audio conference bridges are only available in remote locations; in that situation, enabling this parameter means thatUnified Communications Manager would attempt to use the local video conference bridge first, even for audio-only conference calls. The default value is **False**.
Enable Click-to-Conference for Third-Party Applications	This parameter determines whether the Click-to-Conference functionality over the SIP trunk is enabled on Unified Communications Manager. The Click-to-Conference feature allows third-party applications to setup a conference using the SIP out of dialog REFER method and subscribe to the SIP trunk for Conference Event Package through SIP SUBSCRIBE/NOTIFY. **Warning** Enabling this parameter could negatively affect CTI applications that are not coded to support this feature. Default value: False

Service Parameters	Description
Cluster Conferencing Prefix Identifier	This parameter defines a number, up to 8 digits (e.g. 0001), that is prefixed to a conference identifier generated for Adhoc and Meet-Me conferences that will be hosted on a SIP conference bridge such as Cisco Telepresence MCU or Cisco Telepresence Conductor. This field should be populated by the administrator when there are multiple clusters in a network that will be sharing the SIP conference bridges that Unified Communications Manager manages. Every cluster should be configured with a unique prefix to ensure that the conference identifier for Adhoc and Meet-Me conferences is unique. If conference resources are not being shared across clusters, then this field may not be populated.

Configure Join Across Lines

The Join Across Lines feature allows a user to join calls on multiple phone lines (either on different directory numbers or on the same directory number but on different partitions) to create a conference.

Before you begin

- Ensure the phone model supports Join Across Lines Generate a Phone Feature List, on page 5
- Configure Ad Hoc Conferencing, on page 223

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Default Device Profile**. The **Default Device Profile Configuration** window is displayed.

Step 2 From the **Device Profile Type** drop-down list, choose the phone model.

Step 3 From the **Device Protocol** drop-down list, choose the relevant SCCP or SIP protocol.

Step 4 Set the **Join Across Lines** to **On**.

Step 5 Click **Save**.

Conference Interactions

Feature	Interaction
Conference by Using **cBarge**	Initiate a conference by pressing the **cBarge** softkey, or if the Single Button cBarge feature is enabled, by pressing the shared-line button of the active call. When cBarge is initiated, a barge call gets set up by using the shared conference bridge, if available. The original call gets split and then joined at the conference bridge. The call information for all parties gets changed to Conference. The barged call becomes a conference call with the barge target device as the conference controller. It can add more parties to the conference or can drop any party. When any party releases from the call, leaving only two parties in the conference, the remaining two parties experience a brief interruption and then get reconnected as a point-to-point call, which releases the shared conference resource.
Interaction with Call Park, Call Transfer, and Redirect	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.
Softkey display on SIP phones	The ConfList and the Remove softkey feature is available only on SCCP phones. The SIP phones have a Show Details button with similar functionality.

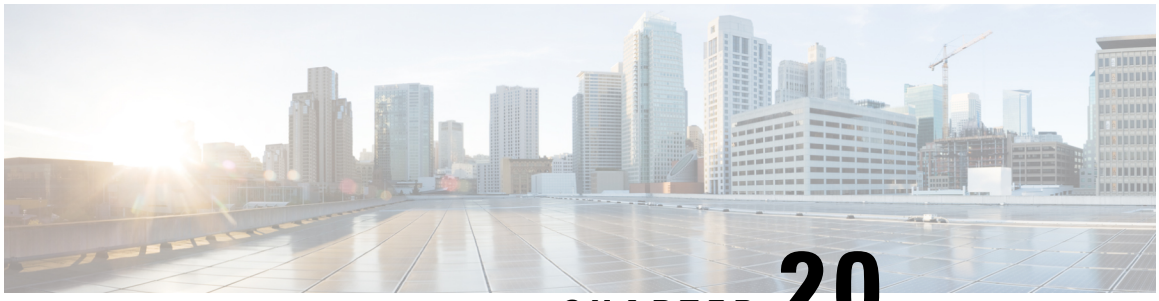
Conference Restrictions

The following restrictions apply to ad hoc conferencing:

Feature	Restrictions
Ad Hoc conference	Unified Communications Manager supports a maximum of 100 simultaneous Ad Hoc conferences for each Unified Communications Manager server. Cisco Unified Communications Manager supports a maximum of 64 participants per Ad Hoc conference (provided adequate conference resources are available). In the case of linked Ad Hoc conferences, the system considers each conference as one participant.

Feature	Restrictions
Ad Hoc conference on SIP phones: • Cisco Unified IP Phone 7911 • Cisco Unified IP Phone 7941 • Cisco Unified IP Phone 7961	Unified Communications Manager uses“ beep” and “beepbeep” 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 “beepbeep”. Users might not hear the beeps because of the time it takes Unified Communications Manager to set up and tear down connections during the conferencing process. You can invoke Ad Hoc conference linking for phones that are running SIP only by using the Conference and Transfer functions. The system does not support Direct Transfer and Join. Supported phones that are running SIP comprise Cisco Unified IP Phone 7911, 7941, 7961.
Ad Hoc conference on SIP phones: • Cisco Unified IP Phone 7940 • Cisco Unified IP Phone 7960 • Third-Party Phone	• Phones display individual calls as conference calls. Cisco Unified IP Phones 7940 and 7960 can create local conference calls but not Ad Hoc conference calls. • Conference list (ConfList), is not available. • Remove last conference participant (RmLstC), is not available. • Drop Ad Hoc conference is not supported. • The SIP Profile parameter Conference Join Enabled controls behavior of the phone that is running SIP when the conference controller exits a locally hosted conference. If the Conference Join Enabled check box is unchecked, all legs disconnect when the conference controller exits the Ad Hoc conference call. If the Conference Join Enabled check box is checked, the remaining two parties stay connected. • To achieve the same level of control that the Drop Ad Hoc Conference parameter settings provide for conference calls that a phone that is running SCCP initiates, the administrator can use a combination of the Conference Join Enabled SIP profile parameter and the Block OffNet to OffNet Transfer service parameter for conferences that are initiated on the phone that is running SIP (Cisco Unified IP Phone 7940 or 60). (Because the phone that is running SIP performs a transfer when it drops out of the conference call, the Block OffNet to OffNet Transfer can prevent toll fraud by not allowing two offnet phones to remain in the call.) • Unified Communications Manager uses “beep” and “beepbeep” 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 “beepbeep”. Users might not hear the beeps because of the time it takes Unified Communications Manager to set up and tear down connections during the conferencing process.

Feature	Restrictions
Phone displaying "To Conference" even when two parties are connected	Configure a Call Manager cluster with Publisher (CmA11) and Subscribers (CmA2). Phones A, B, C are registered with CmA1. Phones D is registered with CmA2. • Setup an consultative or blind ad-hoc conference between A(1000), B(4000), C(5000), D(6000) with A as the controller. • Shutdown Cma2. • Phone D will go to Preservation mode & press end call softkey . • Phone A,B & C are in conference. • Phone A,B & C are in conference. • Disconnect Phone A ,then Phone B & C should be in a Direct call. Issue: Phone B & C are still in conference • Disconnect Phone A ,then Phone B & C should be in a Direct call. Issue: Phone B & C are still in conference • Disconnect Phone B, there should be no call on phone C. Phone B & C are still in conference. Issue: Phone C is still in Conference .



CHAPTER 20

Meet-Me Conferencing

- Meet-Me Conferencing Overview, on page 231
- Meet-Me Conferencing Task Flow, on page 231
- Meet-Me Conferencing Restrictions, on page 236

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

Before you begin

- Refer to the configuration documentation which came with your router and check for any settings which you may need to configure before proceeding with the Meet-Me Conferencing Task Flow.

Procedure

	Command or Action	Purpose
Step 1	Configure a Softkey Template for Meet-Me Conferencing, on page 232	Add the Meet-Me softkey to a softkey template.
Step 2	To Associate a Softkey Template with a Common Device Configuration, on page 233, complete the following subtasks: • Add a Softkey Template to a Common Device Configuration, on page 233 • Associate a Common Device Configuration with a Phone, on page 234	**Optional**. To make the softkey template available to phones, you must complete either this step or the following step.
Step 3	Common Device ConfigurationAssociate a Softkey Template with a Phone, on page 234	**Optional**. Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in

	Command or Action	Purpose
		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 a Meet-Me Conferencing Number, on page 235	Enable advanced conferencing, specify the maximum number of participants, and specify when to drop a conference connection.

Configure a Softkey Template for Meet-Me Conferencing

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

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

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 234.

Before you begin

Configure a Softkey Template for Meet-Me Conferencing, on page 232

Procedure

	Command or Action	Purpose
Step 1	Add a Softkey Template to a Common Device Configuration, on page 233	
Step 2	Associate a Common Device Configuration with a Phone, on page 234	

Add a Softkey Template to a Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.

b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Before you begin

Add a Softkey Template to a Common Device Configuration, on page 233

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey Template with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Before you begin

Configure a Softkey Template for Meet-Me Conferencing, on page 232

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Configure a Meet-Me Conferencing Number

The Cisco Unified Communications Manager administrator provides the Meet-Me conference directory number range to users, so that they can access the feature. The user chooses a directory number from the range that is specified for the Meet-Me Number or Pattern to establish a Meet-Me conference and becomes the conference controller.

Procedure

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

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

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

Step 4 Perform one of the followings tasks:

- To copy a Meet-Me number or pattern, click the Meet-Me number or pattern that you want to copy. The **Meet-Me Number/Pattern Configuration** window appears. Click **Copy**.
- To add a Meet-Me Number or Pattern, click the **Add New** button.
- To update an existing Meet-Me Number or Pattern, click the Meet-Me Number or Pattern that you want to update.

Step 5 Enter the appropriate settings.

See the Related Topics section for information about the fields and their configuration options.

Step 6 Click **Save**.

Meet-Me Number and Pattern Settings

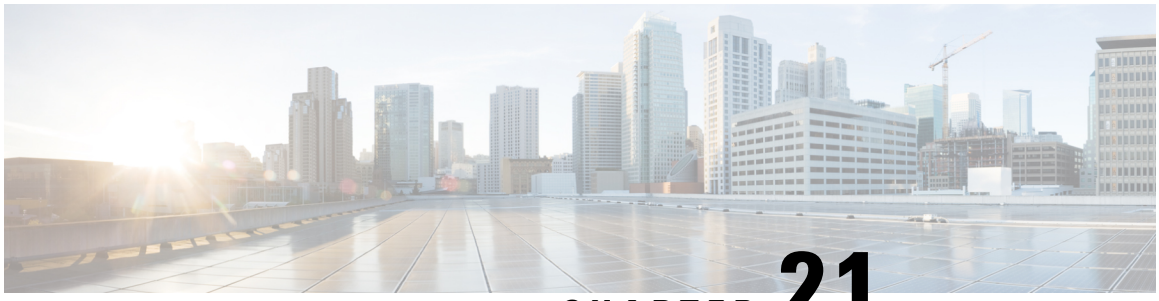
Field	Description
Directory Number or Pattern	Enter a Meet-Me number or a range of numbers. To configure a range, the dash must appear within brackets and follow a digit; for example, to configure the range 1000 to 1050, enter 10[0-5]0.
Description	The description can include up to 50 characters in any language, but it cannot include double quotation marks ("), percentage sign (%), ampersand (&), or angle brackets (<>).

Field	Description
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 Unified Communications Manager cluster.
Minimum Security Level	Choose the minimum Meet-Me conference security level for this Meet-Me number or pattern from the drop-down list box. • Choose **Authenticated** to block participants with nonsecure phones from joining the conference. • Choose **Encrypted** to block participants with authenticated or nonsecure phones from joining the conference. • Choose **Non Secure** to allow all participants to join the conference. **Note** To use this feature, ensure that you have a secure conference bridge that is configured and available.

Meet-Me Conferencing Restrictions

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

After the maximum number of participants that is specified for that conference is has been exceeded, no other callers can join the conference.



CHAPTER 21

Conference Now

- Conference Now Overview, on page 237
- Conference Now Prerequisites, on page 237
- Activate Cisco IP Voice Media Streaming, on page 238
- Configure Conference Now Settings, on page 238
- Enable Conference Now for User, on page 239
- Enable Conference Now via LDAP, on page 239
- Conference Now Interactions, on page 240
- Conference Now Restrictions, on page 241

Conference Now Overview

Conference Now provides a basic audio conferencing solution for small business customers that allows internal and external callers to join a conference via a centralized IVR.

To host a meeting, configured users must configure a meeting PIN that they will need to enter, along with the meeting number, when starting the meeting. The host provides the other meeting participants with the relevant meeting information, including the time slot, meeting number (which is usually the host's primary extension) and an optional access code for secure conferences. At the designated time, the other participants can join the call by dialing the IVR, and entering the meeting information at the prompts.

Administrators must configure end users with the ability to host Conference Now conferences. After the feature is configured, meeting hosts can edit their meeting access code from within the Self-Care Portal.

Note Cisco recommends that you use IPVMS software-based conference bridges for Conference Now. If you use other conference bridges, the conference entry and exit tones may not play to participants.

Conference Now Prerequisites

To use Conference Now you must make sure that the following media resources are configured, and are available to the devices that will be initiating conferences.

- Conference Bridge—For the best user experience, we recommend using a software-based Cisco IPVMS conference bridge. Using another conference bridge might not provide the conference party entry and exit tone.
- Interactive Voice Response (IVR)

After you configure these resources, you can make them available to devices by configuring a media resource group list that includes these resources and then associating that media resource group list to the device pools that will be used by your devices, or to individual devices. For more information on configuring Conference Bridges, Interactive Voice Response, and Media Resource Groups, see "Configure Media Resources" section of the System Configuration Guide for Cisco Unified Communications Manager.

Activate Cisco IP Voice Media Streaming

The Cisco IP Voice Media Streaming Service must be running in order to use IVR services and Conference Now.

Procedure

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

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

Step 3 If the **Cisco IP Voice Media Streaming** application is deactivated, check the corresponding check box and click **Save**.

Configure Conference Now Settings

Use this procedure to configure Conference Now system settings on Unified Communications Manager.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Conference Now**.

Step 2 In the **Conference Now IVR Directory Number** field, enter a DID (Direct Inward Dial) number for a Unified Communications Manager cluster to provide access for external callers.

Step 3 From the Route Partition drop-down list, select a partition.

Note
The combination of the number and the partition must be unique within a cluster.

Step 4 Complete the remaining fields in the **Conference Now Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save**.

What to do next

Enable the feature for end users:

- If you haven't yet synced your LDAP directory, add Conference Now to your LDAP sync, so that newly synced users will be able to host Conference Now meetings. See Enable Conference Now via LDAP, on page 239.
- To enable the feature for an existing end user, see Enable Conference Now for User, on page 239.

Enable Conference Now for User

Use this procedure to configure an existing end user with the ability to host Conference Now meetings.

Note You can also use Bulk Administration's Update Users feature to enable Conference Now for a large number of users via a csv file. You must ensure that the same settings as in the below task are configured. For more information, on how to use Update Users, see Bulk Administration Guide for Cisco Unified Communications Manager.

Procedure

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

Step 2 Click **Find** and select the user for whom you want to add Conference Now.

Step 3 Under **Conference Now**, check the **Enable End User to Host Conference Now** check box.

Step 4 (Optional) For secure conferencing, enter an **Attendees Access Code**. Note that end users will be able to modify their access code setting within the Self-Care Portal.

Note
If the user has a **Self-Service User ID** assigned, the Conference Now **Meeting Number** prepopulates with the value of the **Self-Service User ID**, which defaults to the user's primary extension.

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

Step 6 Click **Save**.

Enable Conference Now via LDAP

If you have not yet synced your LDAP directory, you can enable Conference Now for synced users by adding the option to a feature group template and then adding that feature group template to the initial LDAP sync. New users provisioned via the LDAP sync will have Conference Now enabled.

Note You cannot apply feature group template edits to an LDAP directory sync where the initial sync has already occurred. To apply these edits to an LDAP sync, the initial sync must not yet have occurred.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **User Phone/Add** > **Feature Group Template**.

Step 2 Do either of the following:

- Select **Find** and select an existing template.
- Click **Add New** to create a new template.

Step 3 From the drop-down list, select a **Service Profile**.

Step 4 From the drop-down list, select a **User Profile**.

Step 5 Check the **Enable End User to Host Conference Now** check box.

Step 6 Click **Save**.

What to do next

Assign the template to an LDAP directory sync so that synced users will be configured for Conference Now. For more information on configuring an LDAP sync, see the "Configure End Users" section of the System Configuration Guide for Cisco Unified Communications Manager.

Alternatively, you can add a new user with Conference Now functionality via the **Quick User/Phone Add** menu. You would need to add a new user that uses this feature group template in addition to assigning a primary extension.

Conference Now Interactions

Feature	Interactions
Mobility EFA (Enterprise Feature Access)	A mobility user dials an Enterprise Feature Access DID number from a remote destination. After the call is connected, the remote destination phone is used to send DTMF digits to Unified Communications Manager via the PSTN gateway. The user PIN followed by the # key is first authenticated with Unified Communications Manager. After the user PIN authentication is successful, press 1 and the # key, to indicate a two-stage dialed call, followed by the desired phone number. If the dialed phone number is a Conference Now IVR Directory Number and the user is a meeting host, then the user must enter the PIN again.

Feature	Interactions
Mobility MVA (Mobile Voice Access)	**Important** From Release 15SU4 onwards, Cisco Unified Communications Manager does not support Mobile Voice Access via H.323 and SIP VoiceXML (VXML) Gateways. The Cisco Unified Mobile Voice Access Service is deprecated. A call is directed to Unified Communications Manager through the enterprise PSTN H.323 or SIP gateway. The IVR prompts the user to enter the User ID, # key, PIN, # key, number 1 (to make a Mobile Voice Access call) and then the desired phone number. If the phone number is a Conference Now IVR Directory Number and the user is a meeting host, then the user must enter the PIN again. **Note** Users are not prompted for entering their PIN if they dial directly from their remote destination. However, if they dial from a different phone to Mobile Voice Access Directory Number, then they are prompted to enter PIN before they can make the call. If the users call Conference Now IVR Directory Number, they are prompted to enter the PIN again.

Conference Now Restrictions

The Conference Now feature has the following restrictions:

- The host cannot mute attendees.
- The attendee cannot mute the audio by entering the DTMF digits.
- The list of Conference Now participants is not supported.
- The maximum number of participants in a conference is controlled by the existing CallManager service parameter "Maximum MeetMe Conference Unicast". It applies to both internal and external callers.
- Maximum number of simultaneous Conference Now and MeetMe conference instances combined together is 100 per Unified Communications Manager CallManager node.
- Video on hold is not supported.
- The IPVMS software conference bridge only supports codec G.711 (ALaw & ULaw) and Wide Band 256k. If there is a codec mismatch between the calling device and the software conference bridge, a transcoder will be allocated.
- Ensure that at least one of the following conditions is met to play the conference party entry and exit tone:
  - At least one conference participant is using the Cisco IP Phone.
  - IPVMS is the allocated software conference bridge.
- When the sets up a Conference Bridge, the conference will continue with the remaining attendees irrespective whether the host is present or not. If the host wants to rejoin the conference, an announcement to enter the Attendee Access Code is played if it is configured by the host. The host cannot schedule or mute attendees; therefore, the host status is no longer valid.

- No audio announcement will play if the host is the first person to join the conference. However, when the host dials into Conference Now from an internal IP Phone, there is a visual display on the IP Phone showing “To Conference”.



Note If the host joins the Conference Now from any external phone, then there will be no visual display on the phone.

- When the **Enforce End User PIN Lockout for Conference Now and Mobile Voice Access** service parameter is set to **True** (default), each failed Conference Now host PIN attempt increments the credential hack count for the end-user account associated with the host. Unified Communications Manager locks the end user’s PIN when the number of failures reaches the threshold configured in the assigned Credential Policy.

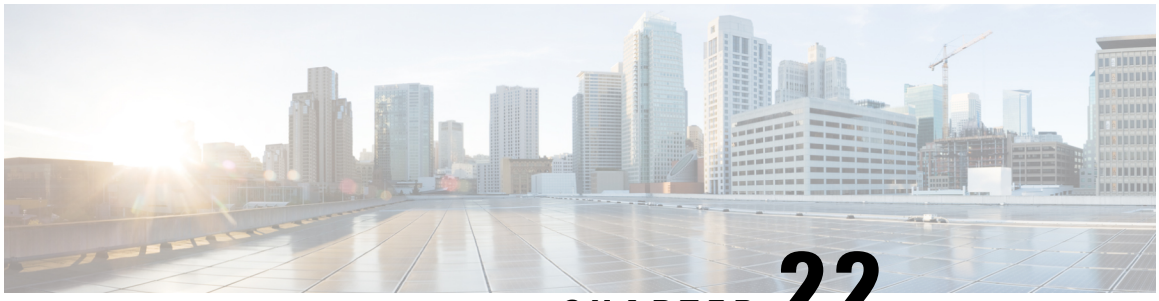
  Conference Now, Mobile Voice Access, Extension Mobility, Personal Directory, and other PIN-authenticated features reject the locked PIN until an administrator unlocks the end user. Setting the parameter to **False** prevents subsequent Conference Now and Mobile Voice Access PIN failures from incrementing the hack count, but it does not unlock an end user who is already locked. We recommend retaining the default value.



PART IX

Placing Calls

- Call Back , on page 245
- Hotline , on page 257
- Speed Dial and Abbreviated Dial, on page 271
- WebDialer , on page 275
- Paging , on page 291
- Intercom , on page 311



CHAPTER 22

Call Back

- Call Back Overview, on page 245
- Call Back Prerequisites, on page 245
- Call Back Configuration Task Flow, on page 246
- Call Back Interactions, on page 251
- Call Back Restrictions, on page 252
- Call Back Troubleshooting, on page 253

Call Back Overview

The CallBack feature allows you to receive notification when a busy extension is available to receive calls.

You can activate Call Back for a destination phone that is within the same Unified Communications Manager cluster as your phone or on a remote Private Integrated Network Exchange (PINX) over QSIG trunks or QSIG-enabled intercluster trunks.

To receive CallBack notification, press the CallBack softkey or feature button while receiving a busy or ringback tone. You can activate Call Back during reorder tone, which is triggered when the No Answer timer expires.

Suspend/Resume

The Call Back feature enables the system to suspend the call completion service if the user who originated Call Back is busy. When the originating user then becomes available, the call completion service resumes for that user.

Note Call Back supports Suspend/Resume CallBack notification for both intracluster and intercluster QSIG trunks or QSIG-enabled intercluster trunks.

Call Back Prerequisites

To use the Call Back feature, the destination phone must be in one of the following locations:

- In the same Unified Communications Manager cluster as the user phone

- On a remote PINX over QSIG trunks
- On a remote PINX over QSIG-enabled intercluster trunks

If you want to use non-English phone locales or country-specific tones, you must install locales.

- The following devices support the Call Back feature:
  - Cisco Unified IP Phones 6900, 7900, 8900, and 9900 Series (except 6901 and 6911)
  - Cisco IP Phones 7800 and 8800 Series
  - Cisco VGC Phone (uses the Cisco VG248 Gateway)
  - Cisco Analog Telephone Adapter (ATA) 186 and 188
  - Busy Subscriber for Cisco VG224 endpoints
  - No Answer for Cisco VG224 endpoints
- A CTI route point that forwards calls to any of the supported phones.

Call Back Configuration Task Flow

Complete one of the task flows depending on whether your phone supports softkey or buttons.

Use this table to determine whether to configure the CallBack softkey or the button for the Call Back supported IP phones.

Table 24: Cisco IP Phones That Use CallBack Softkeys and Buttons

Cisco Phone Model	CallBack Softkey	CallBack Button
Cisco Unified IP Phone 6900 Series (except 6901 and 6911)	X	X
Cisco Unified IP Phone 7900 Series	X	
Cisco IP Phone 7800 and 8800 Series	X	X
Cisco Unified IP Phone 8900 Series	X	X
Cisco Unified IP Phone 9900 Series	X	X
Cisco IP Communicator	X	

Before you begin

- Review Call Back Prerequisites, on page 245.

Procedure

	Command or Action	Purpose
Step 1	Configure Softkey Template for CallBack, on page 247	Perform this step to add CallBack softkey to template and configure the softkey using the Common Device Configuration or phone.
Step 2	Configure CallBack Button, on page 250	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

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one the following procedures:

- Associate CallBack Softkey Template with a Common Device Configuration, on page 248
- Associate CallBack Softkey Template with Phone, on page 249

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 249.

Procedure

	Command or Action	Purpose
Step 1	Add CallBack Softkey Template to the Common Device Configuration, on page 249	Perform this step to add CallBack softkey template to the Common Device Configuration.
Step 2	Associate a Common Device Configuration with a Phone, on page 249	Perform this step to link the CallBack softkey Common Device Configuration to a phone.

Add CallBack Softkey Template to the Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate CallBack Softkey Template with Phone

Optional: Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in conjunction with the Common Device Configuration. Use this procedure in conjunction with the Common Device Configuration if you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.
Step 2 Click **Find** to select the phone to add the softkey template.
Step 3 From the **Softkey Template** drop-down list, choose the template that contains the new softkey.
Step 4 Click **Save**.
Step 5 Press **Reset** to update the phone settings.

Configure CallBack Button

The procedures in this section describe how to configure the CallBack button.

Procedure

	Command or Action	Purpose
Step 1	Configure Phone Button Template for Call Back, on page 250	Perform this step to assign CallBack button features to line or speed dial keys.
Step 2	Associate a Button Template with a Phone, on page 251	Perform this step to configure the CallBack button for a phone.

Configure Phone Button Template for Call Back

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

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Phone Button Template**.
Step 2 Click **Find** to display list of supported phone templates.
Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.
a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.
Step 4 Perform the following steps if you want to add phone buttons to an existing template.
a) Click **Find** and enter the search criteria.
b) Choose an existing template.
Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.
Step 6 Click **Save**.
Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate a Button Template with a Phone

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Call Back Interactions

Feature	Interaction
Call Forward	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.
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 Unified Communications Manager knows when a line on a SIP 7960 or 7940 phone becomes available is by monitoring an incoming SIP INVITE message that Unified Communications Manager receives from the phone. After the phone sends the SIP INVITE to Unified Communications Manager and the phone goes on-hook, Unified Communications Manager sends an audio and CallBack notification screen to the Cisco Unified IP Phone 7960 and 7940 (SIP) user.

Feature	Interaction
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

Feature	Restriction
Call Back with video across CUBE	The Call Back feature does not work for video calls when the call is placed between two Unified CM clusters that are connected via CUBE with qsig-enabled SIP trunks. For additional detail, see CSCun46243.
SIP Trunks	Call Back is not supported over SIP trunks but is supported over QSIG-enabled SIP trunks.
Supported characters for name or number of calling or called party	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 (*).
Voicemail	You cannot activate Call Back if you forward all calls to Voice-Messaging System.

Call Back Troubleshooting

This section describes the problems, possible causes, and solutions for various scenarios, and error messages that are displayed on the IP phone for Call Back.

Unplug/Reset Phone After Pressing CallBack Softkey but Before CallBack Occurs

Problem

You have unplugged or reset the phone after pressing the CallBack Softkey but before activating CallBack.

Possible Cause

Unified Communications Manager cancels the Call Back activation.

Solution

After the caller phone registers, the caller phone does not display the Call Back activation window after the reset. The caller must press the CallBack Softkey to view the active Call Back service. CallBack notification occurs on the phone.

Caller Misses to View Availability Notification Before Phone Reset

Problem

In an intracluster or intercluster Call Back scenario, a caller initiates Call Back for a user, for example, User B, who is unavailable. When User B becomes available, the availability notification screen displays on the caller phone, and a tone plays. The caller misses the availability notification for some reason, and the phone resets.

The caller contacts a different user, User C, for example, and presses the CallBack softkey because User C appears busy. The replace/retain screen displays on the caller phone, but the screen does not state that the availability notification already occurred for User B.

Possible Cause

The user reset the phone.

Solution

After a phone reset but not during an active call, review the Call Back notifications on the phone. Press the CallBack softkey.

Call Back Error Messages

The following section describes the error messages that display on the IP phone screen.

CallBack Is Not Active

Problem

The following error message is displayed:

```
CallBack is not active. Press Exit to quit this screen.
```

Possible Cause

User pressed the CallBack softkey during the idle state.

Solution

Follow the recommended action provided in the error message.

CallBack Is Already Active

Problem

The following error message is displayed:

```
CallBack is already active on xxxx. Press OK to activate on yyyy. Press Exit to quit this
screen.
```

Possible Cause

A user tried to activate Call Back, but it is already active.

Problem

Follow the recommended action provided in the error message.

CallBack Cannot Be Activated

Problem

The following error message is displayed:

```
CallBack cannot be activated for xxxx.
```

Possible Cause

When a user tried to activate Call Back, either the extension is not available in Unified Communications Manager database or there is no QSIG route to the destination (that is, the extension belongs to remote Proxy which is connected via non-QSIG trunk), and the extension is not found in the database.

Solution

The user must try again, or the administrator must add the directory number to the Cisco Unified CM Administration.

Key Not Active

Problem

During a call, the CallBack softkey displays on the phone and the user presses the CallBack softkey before the phone rings. But, the following error message is displayed on the phone:

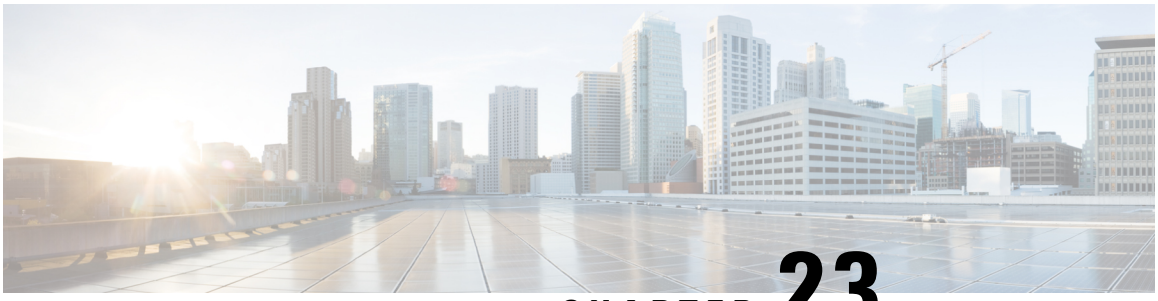
```
Key Not Active
```

Possible Cause

User may not be pressing the CallBack softkey at the appropriate time.

Solution

Users must press the CallBack softkey after a ringing or busy signal is received. Pressing the softkey at the wrong time may cause an error message to display on the phone.



CHAPTER 23

Hotline

- Hotline Overview, on page 257
- System Requirements for Hotline, on page 258
- Hotline Configuration Task Flow, on page 258
- Hotline Troubleshooting, on page 269

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. This is useful for phones that are designated for calling emergency or "hotline" numbers.

The administrator can configure a delay of up to 15-seconds. This allows the user time to place a call before the phone defaults to the hotline number. You can configure the **Off Hook To First Digit Timer** parameter to set the timer in **Device** > **Device Settings** > **SIP Profile**.

Hotline adds the following additional restrictions and administrator controls for phones that use PLAR:

- Hotline devices (devices configured to use hotline) that receive calls will receive calls only from other hotline devices, and will reject non-hotline callers.
- You can configure a Hotline phone to call only, receive only, or both call and receive.
- You can restrict the features available on a Hotline phone by applying a softkey template to the phone.
- Analog hotline phones ignore inbound hookflash signals.

Route Class Signaling

Hotline uses route class signaling to allow Hotline phones to receive calls only from other Hotline phones. A route class is a DSN code that identifies the class of traffic for a call. The route class informs downstream devices about special routing or termination requirements. A Hotline phone can only accept calls from a Hotline phone with the same route class.

Call Screening

Hotline also provides Configurable Call Screening based on caller ID. Configurable Call Screening allows a receiving Hotline phone to screen calls based on caller ID information and allow only callers in a screening list to connect.

System Requirements for Hotline

The following hotline system requirements exist for Unified Communications Manager:

- Unified Communications Manager 8.0(1) or higher on each server in the cluster
- MGCP gateway POTS phones (FXS).
- SCCP gateway POTS phones (FXS).

Tip Cisco Feature Navigator allows you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://cfn.cloudapps.cisco.com/ITDIT/CFN/.

You do not need a Cisco.com account to access Cisco Feature Navigator.

Hotline Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Log in to Cisco Unified Reporting and run a phone feature list report to determine which phones support Hotline.
Step 2	Create Custom Softkey Template, on page 259	Optional. If you want to restrict features on a Hotline phone, create a softkey template that allows only the features that you want.
Step 3	Configure Hotline on Phones, on page 259	Enable the phone as a Hotline device.
Step 4	Configure Route Class Signaling Task Flow, on page 260	Configure route class signaling to support the Hotline feature.
Step 5	Configure Hotline to Call Only or Receive Only Task Flow, on page 264	**Optional**. If you want to restrict a Hotline phone to either originating calls only or terminating calls only, configure call and receive settings.
Step 6	Configure Call Screening with a Calling Search Space , on page 266	**Optional**. Use calling search spaces and partitions to configure a call screening list for your Hotline phones.

Create Custom Softkey Template

When configuring Hotline, you can customize a softkey template to display only those features that you want to make available to a Hotline phone.

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

Before you begin

Generate a Phone Feature List, on page 5

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 redisplays.

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**.

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 259.

Procedure

Step 1 From 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.

Configure Route Class Signaling Task Flow

Perform this task flow to configure route class signaling for Hotline calls.

Procedure

	Command or Action	Purpose
Step 1	Enable Route Class Signaling in the Cluster, on page 261	Set the route class signaling clusterwide defaults for trunks and gateways 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.
Step 2	Enable Route Class Signaling on Trunks, on page 261	Enable route class signaling on an individual trunk.
Step 3	Enable Route Class Signaling on Gateways, on page 262	Enable route class signaling on an MGCP T1/CAS or MGCP PRI gateway.
Step 4	Configure Signaling Labels for the Hotline Route Class, on page 262	Configure SIP signaling labels for Hotline route classes.
Step 5	Configure the Route Class on Hotline Route Patterns, on page 263	Configure the route class on the route patterns that are routing your Hotline calls.
Step 6	Configure the Route Class on Hotline Translation Patterns, on page 263	**Optional**. If you use translation patterns on your Hotline calls, configure the route class on your translation patterns.

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

Step 2 Set the **Route Class Trunk Signaling Enabled** service parameter to **True**.

Step 3 Click **Save**.

What to do next

Use the following procedures to configure route class signaling on individual trunks or gateways.

Enable Route Class Signaling on Trunks, on page 261

Enable Route Class Signaling on Gateways, on page 262

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

Procedure

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

Step 2 Click **Find** and select the SIP trunk on which you want to enable route class signaling.

Step 3 From the **Route Class Signaling Enabled** drop-down list box, choose one of the following options:

- **Default**—This trunk uses the setting from the **Route Class Signaling Enabled** service parameter.
- **Off**—Route class signaling is disabled for this trunk.
- **On**—Route class signaling is enabled for this trunk.

Step 4 Click **Save**.

Enable Route Class Signaling on Gateways

Use this procedure to enable route class signaling on an individual MGCP PRI or MGCP T1/CAS gateway. The configuration for individual gateways overrides the clusterwide service parameter setting.

Before you begin

Follow the Enable Route Class Signaling in the Cluster, on page 261 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 261 procedure to configure route class signaling for individual trunks.

Procedure

Step 1 From 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**.

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 261.

Procedure

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

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

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

Step 4 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 lable to represent the Hotline Data route class. The default value is **ccdata**.

Step 8 Click **Save**.

Configure the Route Class on Hotline Route Patterns

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

For each route pattern that you expect to route a Hotline call, you must set the route class for that route pattern to **Hotline Voice** or **Hotline Data**.

Before you begin

Configure Signaling Labels for the Hotline Route Class, on page 262

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

Procedure

Step 1 From 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**.

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 263 procedure.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Translation Pattern**.

Step 2 Click **Find** to display the translation patterns in your cluster.

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

	Command or Action	Purpose
Step 1	Configure Partitions for Hotline Call Only Receive Only, on page 264	Create two partitions: one should be empty and the other will be assigned to a new CSS.
Step 2	Configure Calling Search Space for Hotline Call Only Receive Only, on page 265	Create a new calling search space and assign one of the new partitions to this CSS. This CSS will contain no other partition.
Step 3	Perform one of the following procedures: • Configure Call Only on Hotline Phone, on page 265 • Configure Receive Only on Hotline Phone, on page 266	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 From 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.

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 264

Procedure

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

Step 2 Click **Add New**.

Step 3 Enter a **Name** and **Description** for the calling search space.

Step 4 From the **Available Partitions** list box, use the arrows to select the **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 265
- Configure Receive Only on Hotline Phone, on page 266

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 265

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Phone**.
Step 2 Click **Find** and select the Hotline phone.
Step 3 From the left navigation pane, click the phone line.
The Directory Number Configuration window displays.
Step 4 From the **Route Partition** drop-down list, select the empty partition that you created.
Step 5 Click **Save**.

Configure Receive Only on Hotline Phone

If you have created your calling search space and partitions already, perform these steps to configure the Hotline phone to receive calls only.

Before you begin

Configure Calling Search Space for Hotline Call Only Receive Only, on page 265

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.
Step 2 Click **Find** and select the Hotline phone.
Step 3 From the **Calling Search Space** drop-down list, select the new 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.

Procedure

	Command or Action	Purpose
Step 1	Configure Partitions for Hotline Call Screening, on page 267	Create any new partitions for your Hotline phone lines.
Step 2	Create Calling Search Space for Hotline Call Screening, on page 268	Create a new CSS for the screening list. The CSS must include partitions with only those Hotline numbers that you want to allow.
Step 3	Configure Hotline Phones for Call Screening, on page 268	Assign the new CSS and partition to the Hotline phone.

Configure Partitions for Hotline Call Screening

To configure call screening in Hotline phones using a calling search space, you must set up partitions where the only Hotline numbers are those that you want to allow.

Perform the following procedure if you need to create a new partition for your Hotline call screening list.

Procedure

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

Step 2 Click **Add New** to create a new partition.

Step 3 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 4 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 5 To create multiple partitions, use one line for each partition entry.

Step 6 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 7 Select one of the following radio buttons to configure the **Time Zone**:

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

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 267

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

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

Step 4 In the **Description** field, enter a description.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

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

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

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

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

Step 8 Click **Save**.

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 268

Procedure

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

Step 2 Click **Find** and select the Hotline phone.

Step 3 From the **Calling Search Space** drop-down list, select the new calling search space that you created for the Hotline call screening list.

Step 4 Click **Save**.

Step 5 From the left navigation pane, click the phone line that you want to use for Hotline calls.
The Directory Number Configuration window displays.

Step 6 From the **Route Partition** drop-down list, select a partition that is included in the calling search space that you set up.

Step 7 Click **Save**.

Hotline Troubleshooting

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

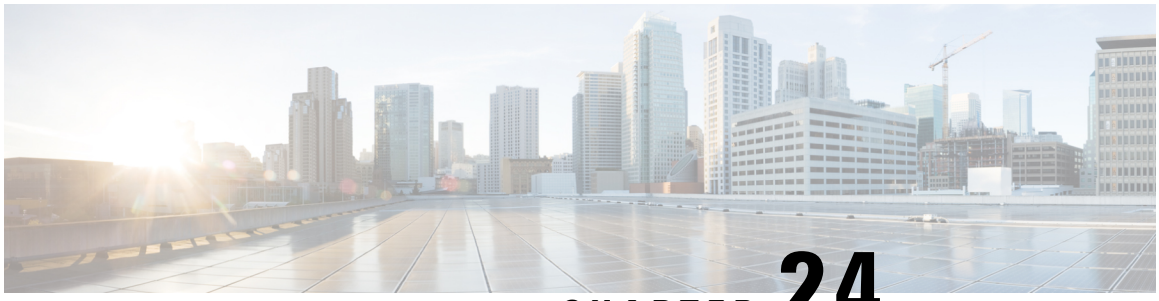
Table 25: Troubleshooting Hotline—Calls Do Not Dial Correctly

Problem	Solution
Dial tone	Check PLAR configuration.
Reorder tone or VCA (intracluster call)	• Check PLAR configuration. • Verify that the phones on both ends are configured as hotline phones.
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.

The following table provides troubleshooting information for cases where call screening based on caller ID does not work.

Table 26: Troubleshooting Hotline—Call Screening Based on Caller ID Problems

Problem	Solution
Call not allowed	• Check Caller ID. • Add pattern to screen CSS.
Call allowed	Remove pattern from screen CSS.



CHAPTER 24

Speed Dial and Abbreviated Dial

- Speed Dial and Abbreviated Dial Overview, on page 271
- Speed Dial and Abbreviated Dial Configuration Task Flow, on page 272

Speed Dial and Abbreviated Dial Overview

Administrators can configure speed dial numbers for phones to provide speed dial buttons for users or to configure phones that do not have a specific user that is assigned to them. Users use the Cisco Unified Communications Self Care Portal to change the speed dial buttons on their phones. When configuring speed dial entries, some of the speed dial entries are assigned to the speed dial buttons on the IP phone; the remaining speed dial entries are used for abbreviated dialing. When a user starts dialing digits, the AbbrDial softkey displays, and the user can access any speed dial entry by entering the appropriate index (code) for abbreviated dialing.

The speed dial settings on the phone are associated with a physical button on a phone, whereas the abbreviated dial settings are not associated with a phone button.

Programming Speed Dials with Pauses

You can program commas in your speed dials to reach destinations that require a Forced Authorization Code (FAC), Client Matter Code (CMC), dialing pause, or additional digits (such as a user extension, meeting access number, or voice mail password). Within a speed dial, each comma (,) represents either:

- A delimiter that separates the destination call address from an FAC or CMC code
- A pause of 2 seconds prior to sending post-connect DTMF digits

For example, let's say that you want a speed dial that includes FAC and CMC codes, followed by IVR prompts where:

- The called number is 91886543.
- The FAC code is 8787.
- The CMC code is 5656.
- The IVR response is 987989#, which must be entered 4 seconds after the call connects.

In this case, you would program **91886543,8787,5656,,987989#** as the speed dial.

Speed Dial and Abbreviated Dial Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the Speed Dial and Abbreviated Dial feature.
Step 2	Configure Speed Dial and Abbreviated Dial, on page 272	Configure Speed Dial and Abbreviated Dial numbers.

Configure Speed Dial and Abbreviated Dial

You can configure a total of 199 speed dial and abbreviated dial settings. Configure speed dial settings for the physical buttons on the phone. Configure abbreviated dial settings for the speed dial numbers that you access with abbreviated dialing. You can configure speed dial entries and abbreviated dial indexes in the same window.

You can also configure post connect DTMF digits as well as FAC , CMC codes as part of the speed dial.

Follow these steps to configure speed dial and abbreviated dial.

Note Not all Cisco Unified IP Phones support abbreviated dialing. See the phone user guide for information.

Before you begin

Generate a Phone Feature List, on page 5

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**. Enter your search criteria and click **Find**. Choose the phone for which you want to configure speed dial buttons.

Step 2 From the **Phone Configuration** window, choose **Add/Update Speed Dials** from the Related Links drop-down list at the top of the window and click **Go**.
The **Speed Dial and Abbreviated Dial Configuration** window appears for the phone.

Step 3 In the **Number** field, enter the number that you want the system to dial when the user presses the speed dial button or the abbreviated dial index for abbreviated dial. You can enter digits 0 through 9, *, #, and +, which is the international escape character. To include dialing pauses in the speed dial, you can enter comma (,) which can act as a delimiter before sending DTMF digits. Each comma you include represents an additional pause of 2 seconds. For example, two commas (,,) represent a pause of 4 seconds. Use of commas also allows you to separate FAC and CMC from the other digits in the speed dial string.

Note

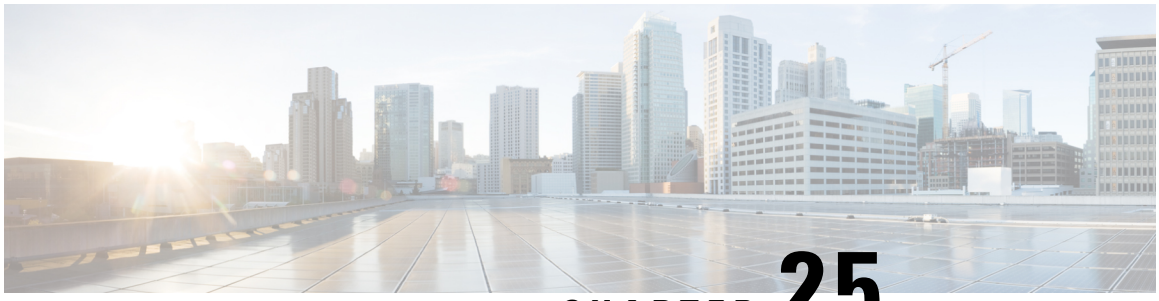
Ensure that the following requirements are met when you include FAC and CMC in the speed dial string:

- FAC must always precede CMC in the speed dial string.
- A speed dial label is required for speed dials with FAC and DTMF digits.
- Only one comma is allowed between FAC and CMC digits in the string.

Step 4 In the **Label** field, Enter the text that you want to display for the speed dial button or abbreviated dial number.

Note
This field is not available for all the phones. To determine whether this field is available for your Cisco Unified IP Phone, see the user documentation for your phone model.

Step 5 (Optional) If you are configuring a pause in speed dial, you must add a label so that FAC, CMC, and DTMF digits are not displayed on the phone screen.



CHAPTER 25

WebDialer

- WebDialer Overview, on page 275
- WebDialer Prerequisites, on page 275
- WebDialer Configuration Task Flow, on page 276
- WebDialer Interactions, on page 286
- WebDialer Restrictions, on page 287
- WebDialer Troubleshooting, on page 287

WebDialer Overview

Cisco WebDialer is installed on a Unified Communications Manager node and used along with Unified Communications Manager. It allows Cisco Unified IP Phone users to make calls from web and desktop applications.

Cisco WebDialer uses hyperlinked telephone numbers in a company directory to allow users to make calls from a web page by clicking on the telephone number of the person that they are trying to call. Cisco WebDialer supports both IPv4 and IPv6 addressing.

In the Cisco Unified Communications Self-Care Portal, from the Directory window, launch Cisco WebDialer using a URL similar to the following:

```
https://<IP address of Cisco Unified Communications Manager server>:8443/webdialer/
Webdialer
```

In the **Cisco WebDialer** screen click **Login** to access the WebdDialer system. A new pop-up window allows you to enter Unified Communications Manager **User ID** and **Password** to perform the necessary Make Call activities.

WebDialer Prerequisites

Cisco WebDialer requires the following software components:

- CTI-supported Cisco Unified IP Phones

WebDialer Configuration Task Flow

Before you begin

- Review WebDialer Prerequisites, on page 275.

Procedure

	Command or Action	Purpose
Step 1	Activate WebDialer, on page 277	Activate the WebDialer service.
Step 2	(Optional) Enable WebDialer Tracing, on page 277	To view WebDialer traces, enable tracing.
Step 3	(Optional) Configure WebDialer Servlet, on page 278	Configure the WebDialer servlet.
Step 4	(Optional) Configure Redirector Servlet, on page 278	If you have multi cluster applications that you develop using HTML over HTTPS interfaces, configure the Redirector servlet.
Step 5	(Optional) Configure WebDialer Application Server, on page 279	To configure Redirector for Cisco WebDialer.
Step 6	(Optional) To Configure Secure TLS Connection to CTI, on page 280, complete the following sub tasks: • Configure WDSecureSysUser Application User, on page 280 • Configure CAPF Profile, on page 178 • Configure Cisco WebDialer Web Service , on page 180	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.
Step 7	Configure Language Locale for WebDialer , on page 283	Determine which language WebDialer displays by setting the locale field in the Cisco Unified Communications Self Care Portal menu.
Step 8	Configure WebDialer Alarms, on page 283	If there are any issues with the Web Dialer feature it alerts the administrator.
Step 9	(Optional) Configure Application Dial Rules, on page 284	If your application requires multiple clusters, configure application dial rules.
Step 10	Add Users to Standard CCM End User Group, on page 284	Add each WebDialer user to the Standard End User Group for Cisco Unified Communications Manager.
Step 11	(Optional) To Configure Proxy User, on page 285, complete the following sub tasks: • Add a WebDialer End User, on page 285	If you use makeCallProxy HTML over HTTP interface to develop an application for using Cisco WebDialer, create a proxy user.

	Command or Action	Purpose
	• Assign Authentication Proxy Rights, on page 286	

Activate WebDialer

Procedure

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

Step 2 From the **Servers** drop-down list, choose the Unified Communications Manager server that is listed.

Step 3 From **CTI Services**, check the **Cisco WebDialer Web Service** check box.

Step 4 Click **Save**.

Step 5 From Cisco Unified Serviceability, choose **Tools** > **Control Center - Feature Services** to confirm that the CTI Manager service is active and is in start mode.

For WebDialer to function properly, the CTI Manager service must be active and in start mode.

What to do next

Configure Language Locale for WebDialer , on page 283 or complete any or all of the following optional tasks:

- Enable WebDialer Tracing, on page 277
- Configure WebDialer Servlet, on page 278
- Configure Redirector Servlet, on page 278
- Configure WebDialer Application Server, on page 279
- Configure Secure TLS Connection to CTI, on page 280

Enable WebDialer Tracing

To enable Cisco WebDialer tracing, use the Cisco Unified Serviceability Administration application. Trace settings apply to both the WebDialer and Redirector servlets. To collect traces, use the Real Time Monitoring Tool (RTMT).

To access the WebDialer trace files, use the following CLI commands:

- **file get activelog tomcat/logs/webdialer/log4j**
- **file get activelog tomcat/logs/redirector/log4j**

For more information about traces, see the *Cisco Unified Serviceability Administration Guide*.

Before you begin

Activate WebDialer, on page 277

Procedure

Step 1 From the navigation drop-down list of the Cisco Unified Communications Manager application, choose **Cisco Unified Serviceability** and then click **Go**.

Step 2 Choose **Trace** > **Configuration**.

Step 3 From the **Server** drop-down list, choose the server on which to enable tracing.

Step 4 From the **Service Group** drop-down list, choose CTI Services.

Step 5 From the **Service** drop-down list, choose the **Cisco WebDialer Web Service**.

Step 6 In the **Trace Configuration** window, change the trace settings according to your troubleshooting requirements.

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

Step 7 Click **Save**.

Configure WebDialer Servlet

The WebDialer servlet is a Java servlet that allows Cisco Unified Communications Manager users in a specific cluster to make and complete calls.

Before you begin

Activate WebDialer, on page 277

Procedure

Step 1 Choose **System** > **Service Parameters**.

Step 2 From the **Server** drop-down list, choose the Cisco Unified Communications Manager server on which to configure Cisco WebDialer web service parameters.

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

Step 4 Configure the relevant WebDialer Web Service parameters. For detailed information about the parameters, see online help.

Step 5 Restart the Cisco WebDialer Web Service for new parameter values to take effect.

Configure Redirector Servlet

The 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.

Note Ensure that all the nodes set up for inter-cluster HTTPS communication in Unified Communications Manager supports the minimum TLS version configured.

Before you begin

Activate WebDialer, on page 277

Procedure

Step 1 From Cisco Unified 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*.

Configure WebDialer Application Server

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

Before you begin

Activate WebDialer, on page 277

Procedure

Step 1 From Cisco Unified Communications Manager Administration Application server window, choose **System** > **Application Server**.

Step 2 From the **Application Server Type** drop-down list, choose a **Cisco WebDialer application server**.
The server appears in the **List of WebDialers** field in the **Service Parameter Configuration** window for the Cisco WebDialer Web Service.

Configure Secure TLS Connection to CTI

WebDialer uses WDSecureSysUser application user credentials to establish a secure TLS connection to CTI to make calls. To configure the WDSecureSysUser application user to establish a secure TLS connection, complete the following tasks.

Before you begin

- Install and configure the Cisco CTL Client. For more information about CTL Client, see Security Guide for Cisco Unified Communications Manager .
- Verify that the Cluster Security Mode in the Enterprise Parameters Configuration window is 1 (mixed mode). Operating the system in mixed mode impacts other security functions in your system. If your system is not currently running in mixed mode, do not switch to mixed mode until you understand these interactions. For more information, see Security Guide for Cisco Unified Communications Manager.
- Verify that the Cluster SIPOAuth Mode field is set to Enabled.
- Activate the Cisco Certificate Authority Proxy Function service on the first node.
- Activate WebDialer, on page 277

Procedure

	Command or Action	Purpose
Step 1	Configure WDSecureSysUser Application User, on page 280	Configure a WDSecureSysUser application user.
Step 2	Configure CAPF Profile, on page 178	Configure a CAPF profile for the WDSecureSysUser application user.
Step 3	Configure Cisco WebDialer Web Service , on page 180	Configure service parameters for the Cisco WebDialer Web service.

Configure WDSecureSysUser Application User

Procedure

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

Step 2 Click **Find**.

Step 3 From the **Find and List Application Users Application** window, choose **WDSecureSysUser**.

Step 4 Configure the fields in the **Application User Configuration** window and click **Save**.

What to do next

Configure CAPF Profile, on page 178

Configure CAPF Profile

Certificate Authority Proxy Function (CAPF) is a component that performs tasks to issue and authenticate security certificates. When you create an application user CAPF profile, the profile uses the configuration details to open secure connections for the application.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **Application User CAPF Profile**.

Step 2 Perform one of the following tasks:

- To add a new CAPF profile, click **Add New** in the **Find** window.
- To copy an existing profile, locate the appropriate profile and click the **Copy** icon for that record in the **Copy** column.

To update an existing entry, locate and display the appropriate profile.

Step 3 Configure or update the relevant CAPF profile fields. See the Related Topics section information about the fields and their configuration options.

Step 4 Click **Save**.

Step 5 Repeat the procedure for each application and end user that you want to use security.

CAPF Profile Settings

Setting	Description
Application User	From the drop-down list, choose the application user for the CAPF operation. This set This setting does not appear in the **End User CAPF Profile** window.
End User ID	From the drop-down list, choose the end user for the CAPF operation. This setting dis This setting does not appear in the **Application User CAPF Profile** window.
Instance ID	Enter 1 to 128 alphanumeric characters (a-z, A-Z, 0-9). The Instance ID identifies the You can configure multiple connections (instances) of an application. To secure the co instance that runs on the application PC (for end users) or server (for application users This field relates to the CAPF Profile Instance ID for Secure Connection to CTIMana
Certificate Operation	From the drop-down list, choose one of the following options: • **No Pending Operation**—This message is displayed when no certificate operatio • **Install/Upgrade**—This option installs a new certificate or upgrades an existing l
Authentication Mode	The authentication mode for the Install/Upgrade certificate operation specifies By Autho a locally significant certificate only when the user or administrator enters the CAPF au

Setting	Description
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 authenti This string supports one-time use only; after you use the string for the instance, you canno
Generate String	To automatically generate an authentication string, click this button. The 4- to10-digit aut
Key Size (bits)	From the drop-down list, choose the key size for the certificate. The default setting is 102 Key generation, which is set at low priority, allows the application to function while the a
Operation Completes by	This field, which supports all certificate operations, specifies the date and time by which The values that are displayed apply for the first node. Use this setting with the **CAPF Operation Expires in (days)** enterprise parameter, which s 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 suc You cannot change the information that is displayed in this field.

Configure Cisco IP Manager Assistant

Procedure

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

Step 2 From the **Server** drop-down list, choose the server on which the Cisco WebDialer Web service is active.

Step 3 From the **Service** drop-down list, choose the **Cisco WebDialer Web** service.
A list of parameters appears.

Step 4 Navigate to and update the CTIManager Connection Security Flag and CAPF Profile Instance ID for Secure Connection to CTIManager parameters.

To view parameter descriptions, click the parameter name link.

Note
CTIManager supports IPv4 and IPv6 addresses.

Step 5 Click **Save**.

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

What to do next

Refer to the Manager Assistant Task Flow for Shared Lines, on page 170 to determine the next task to complete.

Configure Language Locale for WebDialer

Use the Cisco Unified Communications Self Care Portal to configure a language locale for Cisco WebDialer. The default language is English.

Before you begin

Activate WebDialer, on page 277

Procedure

Step 1 From the Cisco Unified Communications Self Care Portal, click the **General Settings** tab.

Step 2 Click **Language**.

Step 3 From the **Display Language** drop-down list, select a language local, and then click **Save**.

Configure WebDialer Alarms

Cisco WebDialer service uses Cisco Tomcat to generate alarms.

Before you begin

Configure Language Locale for WebDialer , on page 283

Procedure

Step 1 From Cisco Unified Serviceability, choose **Alarm** > **Configuration**.

Step 2 From the **Server** drop-down list, choose the server on which to configure the alarm and then click **Go**.

Step 3 From the **Services Group** drop-down list, choose **Platform Services** and then click **Go**.

Step 4 From the **Services** drop-down list, choose **Cisco Tomcat** and then click **Go**.

Step 5 If your configuration supports clusters, check the **Apply to All Nodes** check box to apply the alarm configuration to all nodes in the cluster.

Step 6 Configure the settings, as described in Alarm configuration settings, which includes descriptions for monitors and event levels.

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

Step 7 Click **Save**.

What to do next

Add Users to Standard CCM End User Group, on page 284 or (optionally) if your application requires multiple clusters, see Configure Application Dial Rules, on page 284.

Configure Application Dial Rules

Before you begin

Configure WebDialer Alarms, on page 283

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 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 Unified Communications Manager, you must add each user to the Standard Unified Communications Manager End Users Group.

Procedure

Step 1 Choose **User Management** > **User Group**.

Step 2 In the **Find and List User Group** window, click **Find**.

Step 3 Click **Standard CCM End Users**.

Step 4 In the **User Group Configuration** window, click **Add End Users to Group**.

Step 5 In the **Find and List Users** window, click **Find**. You can enter criteria for a specific user.

Step 6 To add one or more users to the user group, complete one of the following steps:

- To add one or more users, check the check box beside each user to add and then click **Add Selected**.
- To add all users, click **Select All** and then click **Add Selected**.

The users appear in the Users in Group table of the **User Group Configuration** window.

Configure Proxy User

If you use makeCallProxy HTML over HTTP interface to develop an application for using Cisco WebDialer, create a proxy user. For information about the makeCallProxy interface, see the makeCallProxy section in the *Cisco WebDialer API Reference Guide*.

Note MakeCallProxy HTTP Methods is a service parameter under WebDialer Service. This parameter controls the HTTP methods that the MakeCallProxy API accepts. HTTP GET is considered insecure because the credentials required by the API are included as parameters in HTTP GET requests. Hence these HTTP GET parameters can be captured in the application logs and in the web browser's history.

When the service parameter MakeCallProxy HTTP Methods is set to Secure, request made by the HTTP GET will be rejected. By default the parameter MakeCallProxy HTTP Methods is set to Insecure, so that the API accepts both GET and POST methods and the backward compatibility is maintained.

Before you begin

Add Users to Standard CCM End User Group, on page 284

Procedure

	Command or Action	Purpose
Step 1	(Optional) Add a WebDialer End User, on page 285	Add a new user. If the user exists, you can proceed to the next task.
Step 2	Assign Authentication Proxy Rights, on page 286	Assign authentication proxy rights to an end user.

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

Step 7 Click **Save**.

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

Feature	Interaction
Client Matter Codes (CMC)	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.
Forced Authorization Codes (FAC)	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.
ApplicationDialRule table	Cisco WebDialer uses change notifications on the ApplicationDialRule database table to track and use updated dial rules.

Feature	Interaction
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

Feature	Restrictions
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 Web Dialer 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 Unified Communications ManagerCisco 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 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 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 Unified Communications Manager.

Destination Not Reachable

Problem

Cisco WebDialer displays the following message on the End Call window:

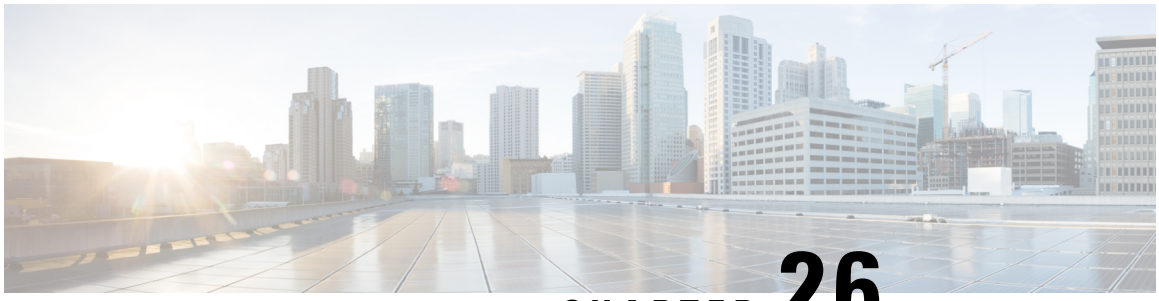
Destination not reachable.

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.



CHAPTER 26

Paging

- Paging Overview, on page 291
- Paging Prerequisites, on page 292
- Cisco Unified Communications Manager Configuration for Basic Paging Task Flow, on page 293
- Advanced Notification Paging Configuration Task Flow, on page 303
- Paging Interactions, on page 310

Paging Overview

Unified Communications Manager can be configured to integrate with Cisco Paging Server to provide basic paging services for Cisco Unified IP Phone and a variety of endpoints. The Cisco Paging Server product is offered through the InformaCast Virtual Appliance and offers the following deployment options:

InformaCast Basic Paging

InformaCast Basic Paging provides phone-to-phone live audio paging to individual Cisco IP phones or groups of up to 50 phones simultaneously. InformaCast Basic Paging is free to all Unified Communications Manager customers and all Cisco Business Edition 6000 and Cisco Business Edition 7000 customers.

InformaCast Advanced Notification

InformaCast Advanced Notification is a full-featured emergency notification and paging solution that allows you to reach an unlimited number of Cisco IP phones and various devices and systems with text and audio messages.

To streamline the configuration process, Unified Communications Manager comes with a provisioning wizard that allows you to quickly configure advanced notifications services.

Some of the features include:

- Text and audio (live or pre-recorded) to Cisco IP Phones and other endpoints
- Analog and IP overhead paging systems integration
- 911 or emergency call monitoring or alerting or recording
- Cisco Jabber integration
- Cisco Spark integration

- Automated weather notifications
- Dynamically triggered emergency conference calls
- Pre-recorded or scheduled broadcasts (school bells or shift changes)
- Event accountability with message confirmation and reporting
- Notification to computer desktops (Windows and Mac OS)
- Facilities integration (control lighting, door locks)
- Security integration (panic or duress buttons, motion detectors, fire)

Purchase a license key to access InformaCast Advanced Notification features.

InformaCast Mobile

InformaCast Mobile is a cloud-based service that allows users to send images, text, and pre-recorded audio to mobile devices running iOS or Android. It also has bi-directional integration with InformaCast Advanced Notification.

Some of the features include:

- The ability to send and receive InformaCast messages via mobile devices running iOS or Android
- Bi-directional integration with InformaCast Advanced Notification
- Message confirmations and read receipts
- No calling or SMS messaging fees

InformaCast Mobile must be purchased direct from Singlewire Software. Please refer to the Singlewire website for addtional details and downloads.

If you have already configured Unified Communications Manager to integrate with InformaCast Advanced Notification, no further configuration of Unified Communications Manager is required.

Paging Prerequisites

Cisco Paging Server is designed to work in a multicast environment. You must configure your network for multicast.

For a list of Cisco Unified IP Phones that support paging, refer to the **Cisco Unified IP Phones** section of the Singlewire Compatibility Matrix at:

http://www.singlewire.com/compatibility-matrix.html.

Cisco Unified Communications Manager Configuration for Basic Paging Task Flow

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

Before you begin

- Learn more about the feature by reviewing the following:
  - Paging Overview, on page 291
  - InformaCast Basic Paging, on page 291
- Review Paging Prerequisites, on page 292
- The configuration in this section is automated when using the Emergency Notifications Paging wizard.

Procedure

	Command or Action	Purpose
Step 1	Enable SNMP Service, on page 294	Configure SNMP in Unified Communications Manager.
Step 2	Set Default Codec to G.711, on page 295	Set the default codec to G.711.
Step 3	Configure a Device Pool for Paging, on page 296	Configure a device pool.
Step 4	Configure Route Partition for InformaCast Paging, on page 297	Configure a route partition for Basic Paging.
Step 5	Configure Calling Search Space for InformaCast Paging, on page 297	Configure a calling search space for Basic Paging.
Step 6	Configure CTI Ports for Paging, on page 298	Configure CTI ports.
Step 7	Configure Access Control Group with AXL Access, on page 298	Configure an AXL access control group.
Step 8	Configure Application User for Paging, on page 299	Configure an application user.
Step 9	Enable web access for the phone using one of the following procedures: • Enable Web Access for a Phone, on page 300 • Enable Web Access for Common Phone Profile, on page 300	You can enable web access on all phones globally using Enterprise Phone Configuration, a group of phones using a Common Phone Profile, or an individual phone.

	Command or Action	Purpose
	• Enable Web Access for Enterprise Phone Configuration, on page 301	
Step 10	Configure Authentication URL, on page 301	Configure the Unified Communications Manager authentication URL to point to InformaCast so that when InformaCast pushes broadcasts to Cisco Unified IP Phone, the phones will authenticate with InformaCast.

For detailed procedures on how to configure Cisco Unified Communications Manager and Cisco Paging Server, refer to the *InformaCast Virtual Appliance Basic Paging Installation and User Guide* .

Configure SNMP for Paging

Perform the following tasks to configure SNMP services in the cluster.

Procedure

	Command or Action	Purpose
Step 1	Enable SNMP Service, on page 294	Enable the SNMP and other services in the cluster.
Step 2	Create an InformaCast SNMP Community String, on page 295	Configure an SNMP community string.

Enable SNMP Service

To configure paging, you must enable SNMP on every node in the cluster. In addition, you must enable the following services:

- Cisco CallManager SNMP Service—Enable on all nodes in the cluster.
- Cisco CallManager—Enable on at least one node.
- Cisco AXL Web Services—Enable on at least one node.
- Cisco CTIManager—Enable on at least one node.

Procedure

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

Step 2 From the **Server** drop-down list, choose the server on which you want to configure SNMP.

Step 3 Check the check boxes that correspond to the **Cisco CallManager SNMP Service**.

Step 4 For at least one server in the cluster, check the check boxes that correspond to **Cisco CallManager**, **Cisco CTIManager**, and **Cisco AXL Web Service** services.

Step 5 Click **Save**.

Step 6 Click **OK**.

Step 7 Repeat the previous steps for all nodes in the cluster.

Create an InformaCast SNMP Community String

Perform this procedure for Basic Paging to set up an SNMP community string.

Before you begin

Enable SNMP Service, on page 294

Procedure

Step 1 From Cisco Unified Serviceability, choose **SNMP** > **V1/V2c** > **Community String**.

Step 2 From the **Server** drop-down list, 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 list, 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

Set Default Codec to G.711, on page 295

Configure Region for Paging

For Basic Paging, you must set up a region for your paging deployment.

Procedure

	Command or Action	Purpose
Step 1	Set Default Codec to G.711, on page 295	Create a region that uses the G.711 codec for calls to other regions.
Step 2	Configure a Device Pool for Paging, on page 296	Set up a device pool for paging and assign the region that you created to that device pool.

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 294

Procedure

Step 1 From 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, 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**.

Configure a Device Pool for Paging

Perform this procedure to configure a device pool for your paging deployment.

Before you begin

Set Default Codec to G.711, on page 295

Procedure

Step 1 From 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, 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, 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, choose **ICVA**.
Step 7 From the **SRST Reference** drop-down list, select **Disable**.
Step 8 Click **Save**.

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.

Procedure

	Command or Action	Purpose
Step 1	Configure Route Partition for InformaCast Paging, on page 297	Configure a route partition for InformaCast paging.
Step 2	Configure Calling Search Space for InformaCast Paging, on page 297	Configure a calling search space for InformaCast paging.

Configure Route Partition for InformaCast Paging

Create a route partition for InformaCast paging.

Before you begin

Configure a Device Pool for Paging, on page 296

Procedure

Step 1 From 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.

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter `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**.

Configure CTI Ports for Paging

Perform this procedure to configure CTI ports for your paging deployment. The number of CTI ports that you need depends on your deployment type and your applications' usage:

- For Basic Paging deployments, you must create a minimum of two CTI ports for InformaCast paging.

Before you begin

Configure Calling Search Space for InformaCast Paging, on page 297

Procedure

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

Step 2 Click **Add New**.

Step 3 From the **Phone Type** drop-down list, choose **CTI Port**.

Step 4 In the **Device Name** field, enter a name for the CTI Port. For example, `ICVA-IC-001` for an InformaCast port.

Step 5 In the **Description** field, enter a description for the port. For example, `InformaCast Recording Port` for Call Monitoring.

Step 6 From the **Device Pool** drop-down list, select **ICVA**.

Step 7 From the **Calling Search Space** drop-down list, select **ICVA**.

Step 8 From the **Device Security Profile** drop-down list, 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, select the following ports:

- For InformaCast ports, select **ICVA-CTIOutbound**.

Step 14 In the **Display (Internal Caller ID)** text box, enter `InformaCast`.

Step 15 In the **ASCII Display (Internal Caller ID)** text box, enter `InformaCast`.

Step 16 Click **Save**.

Step 17 Repeat this procedure for each CTI port that you need.

What to do next

Configure Access Control Group with AXL Access

Perform this procedure to create an access control group that includes AXL access.

Procedure

Step 1 From 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 list, 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**.

Configure Application User for Paging

Perform this procedure to configure an application user:

- For Basic Paging, configure an InformaCast application user.

Procedure

Step 1 From 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`.

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**.

Step 7 Click **Find**.

Step 8 Check the following check boxes (unless otherwise indicated, select these permissions for all application users):

- ICVA User Group
- Standard CTI Allow Control of All Devices
- Standard CTI Allow Control of Phones supporting Connected Xfer and conf
- Standard CTI Allow Control of Phones supporting Rollover Mode
- Standard CTI Enabled

Step 9 Click **Add Selected**.

Step 10 Click **Save**.

Enable Web Access for a Phone

Perform this procedure in Basic Paging to enable web access for a Cisco Unified IP Phone. You can also use a Common Phone Profile to enable web access for a group of phones that use that profile. For details, see Enable Web Access for Common Phone Profile, on page 300.

Before you begin

Configure Application User for Paging, on page 299

Procedure

Step 1 From 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 list, select **Enabled**.
Step 4 Click **Save**.

What to do next

Configure Authentication URL, on page 301

Enable Web Access for Common Phone Profile

Perform this procedure in Basic Paging to enable web access for a group of Cisco Unified IP Phones that use a Common Phone Profile. You can also enable web access on an individual phone. For details, see Enable Web Access for a Phone, on page 300.

Before you begin

Configure Application User for Paging, on page 299

Procedure

Step 1 From Cisco Unified CM Administration, chooseI **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

Configure Authentication URL, on page 301

Enable Web Access for Enterprise Phone Configuration

Perform this procedure in Unified Communications Manager to enable web access for a group of Cisco Unified IP Phone that use a Common Phone Profile. You can also enable web access on an individual phone. For more details, see Enable Web Access for a Phone, on page 300.

Before you begin

Configure Application User for Paging, on page 299.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Enterprise Phone Configuration**.
Step 2 From the **Web Access** drop-down list, select **Enable**.
Step 3 Click **Save**.
Step 4 Click **Apply Config** to reset the phones that use the Common Phone Profile.
Step 5 Click **OK**.

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 Unified Communications Manager.

Procedure

	Command or Action	Purpose
Step 1	Set Authentication URL, on page 301	Set the Unified Communications Manager authentication URL to point InformaCast.
Step 2	Reset Your Phones, on page 302	Reset the phones in your deployment so that your phones use the new settings.
Step 3	Test Your Phones, on page 302	Verify that the phones in your deployment use the new authentication URL settings.

Set Authentication URL

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

Procedure

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

Step 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.

Step 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.

Step 4 Click **Save**.

Reset Your Phones

After you set the authentication URL to point to the InformaCast Virtual Applicance, 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.

Procedure

Step 1 From 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**.

Test Your Phones

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

Procedure

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

Step 2 Use the drop-down list 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.

Advanced Notification Paging Configuration Task Flow

Perform the following tasks to integrate InformaCast Paging Server with Unified Communications Manager for IP paging and emergency call alerting. It includes the following features:

- InformaCast advanced notification
- Panic button configuration
- Text and audio notification to IP phones when a user dials an emergency services number (CallAware)

Procedure

	Command or Action	Purpose
Step 1	Install the InformaCast Virtual Appliance, on page 303.	Download the InformaCast OVA file from the Singlewire website and upload it to vSphere.
Step 2	Configure Connection to InformaCast, on page 305.	Configure Unified Communications Manager and InformaCast.
Step 3	Configure Panic Button, on page 307.	Configure a panic button to send a text and audio notification to IP phones.
Step 4	Configure CallAware Emergency Call Alerting, on page 308.	Configure emergency call text and audio notifications.

Install the InformaCast Virtual Appliance

Singlewire supports InformaCast Virtual Appliance on the VMware ESXi platform, which is managed through the vSphere client.

Note To view a list of Singlewire-supported VMware ESXi versions, go to this URL: https://www.singlewire.com/compatibility-matrix and click the Server Platforms link under InformaCast Platform section.

Note If you have purchased a license, refer to https://www.singlewire.com/icva-kb-activate to activate your license. This will ensure that Emergency Notifications stay active after the 90-day trial.

Note For more details on the installation, including InformaCast screen captures, go to this URL: https://www.singlewire.com/icva-kb-install.

Before you begin

Import InformaCast Virtual Appliance using the vSphere client. This can be downloaded from your VMware server.

Procedure

Step 1 Download the OVA file from the Singlewire website and then log in to the vSphere client.

Note
If you are using InformaCast on the Communications Manager Business Edition 6000, you are supplied with a DVD in a package with an OVA on it (physical media).

The **vSphere Client** window appears.

Step 2 From the **vSphere Client** window, choose **File** > **Deploy OVF Template**.
The **Deploy OVF Template** dialog box appears.

Step 3 Click the **Deploy from File** radio button and then click **Browse** to select the saved the OVA file (or to the OVA file on the supplied DVD). After you select the OVA file, click **Open**.
The **Source** location is selected in the **Deploy OVF Template** dialog box.

Step 4 Click **Next** to continue.
The **Deploy OVF Template** dialog box refreshes and **OVF Template Details** appears.

Step 5 Click **Next** to verify the **Name and Location**, and then click **Next** to select the network to store the new virtual machine files.

Tip
It is good practice to place the Virtual Appliance on the same VLAN as your Cisco Unified Communications Manager.

Step 6 Click **Next** to continue, and then click **Finish**.
The InformaCast Virtual Appliance begins importing.

Step 7 From the **vSphere Client** window, click **Hosts and Clusters** icon and then select your host server.
The **vSphere Client** window refreshes.

Step 8 Click the **Configuration** tab and select the **Virtual Machine Startup/Shutdown** link in the **Software** section.

Step 9 Click the **Properties** link.
The **Virtual Machine Startup and Shutdown** dialog box appears.

Step 10 Check the **Allow virtual machines to start and stop automatically with the system** check box under **System Settings**.

Step 11 Under **Startup Order**, scroll to the **Manual Startup** section and select your virtual machine (by default, this is Singlewire InformaCast VM), and then move it from the **Manual Startup** section to the **Automatic Startup** section, by using the **Move Up** button. After moving it, click **OK**.
The InformaCast Virtual Appliance starts and stops automatically with the server on which it is hosted. Now you can turn on InformaCast's virtual machine and set its network configuration.

Step 12 Choose **View** > **Inventory** > **VMs and Templates** and then select your virtual machine.

Step 13 Choose the **Inventory** > **Virtual Machine** > **Open Console**
The Singlewire InformaCast VM console window appears.

Step 14 InformaCast configuration starts for the first time. During this configuration, perform the following tasks for the InformaCast Virtual Appliance:

a) Accept Cisco End User License Agreement (EULA)
b) Accept Singlewire EULA
c) Set up hostname
d) Set up IP address, subnet mask, and default gateway
e) Set up DNS server IP address and domain name
f) Set up NTP server IP address or hostname
g) Set up time zone
h) Set up Secure Socket Layer (SSL) certificate parameters
i) Set up SSL subject alternate names (optional)
j) Set up the OS admin password
k) Set up the InformaCast and PTT (PushToTalk) admin password. This password is required to connect the Cisco Unified Communications Manager and InformaCast in the Cisco Unified CM Administration, **Advanced Features** > **Emergency Notifications Paging**.
l) Set up security passphrase for backup and communication

When your configuration is successful, the “Welcome to Singlewire InformaCast” message is displayed.

Step 15 Click **Continue** to work with Singlewire InformaCast.

Configure Connection to InformaCast

Use this procedure to load the InformaCast certificate to the Unified Communications Manager Tomcat trust store.

Note Ensure that your InformaCast server supports the minimum TLS version configured on Unified CM.

Before you begin

Install the InformaCast Virtual Appliance, on page 303.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Advanced Features** > **Emergency Notifications Paging**.

Step 2 In the **Introduction to InformaCast Emergency Notifications** page, click **Next** to continue.
The **Installing the InformaCast Virtual Appliance** page appears.

Step 3 In the **Installing the InformaCast Virtual Appliance** page, click **Next** to continue.

Note
You should have successfully installed InformaCast Virtual Appliance to configure with the Unified Communications Manager.

The **Connecting Cisco Unified Communications Manager and InformaCast** page appears.

Step 4 In the **IP address of InformaCast VM** field, enter either IP address or Hostname.

Note
By default, the username is stated as `admin` in the **Username to use in InformaCast** field, and it is not editable.

Step 5 In the **Password for admin app user** field, enter the administrator password of the InformaCast application.
The dialog box displaying the thumbprint of the InformaCast certificate is displayed.

Step 6 Click **OK** to load the InformaCast certificate to the Unified Communications Manager Tomcat trust store.
The configuration process starts.

Note
When the configuration is successful, the **Status** field displays the completion status.

Step 7 Click **Next**.
The wizard performs the following tasks:

- Activates SNMP service
- Configures SNMP Service with locally generated random credentials
- Activates CTI Manager Service
- Configures Unified Communications Manager for InformaCast
    - Creates new region (1 per cluster)
    - Creates new device pool (1 per cluster)
    - Creates SIP trunk (1 per cluster)
    - Creates route group (1 per cluster)
    - Creates route list
    - Creates role
    - Creates app user
- Configures InformaCast for Unified Communications Manager
    - Creates a cluster

- Refreshes recipient groups
- Sets SIP access to deny
- Creates SIP access

Configure Panic Button

Use this procedure to configure a panic button to send a text and audio notification to IP phones. This allows you to initiate a one click alarm if there is emergency.

Before you begin

Configure Connection to InformaCast, on page 305.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Advanced Features** > **Emergency Notifications Paging**.

Step 2 In the **Introduction to InformaCast Emergency Notifications** page, click **Next** to continue.

Step 3 In the **Installing the InformaCast Virtual Appliance** page, click **Next** to continue.

Step 4 In the **Connecting Cisco Unified Communications Manager and InformaCast** page, click **Next** to continue.
The **Configuring a Panic Button** page appears.

Step 5 From the **Choose pre-recorded message by name** drop-down list, select the pre-recorded message to be displayed on Cisco Unified IP phones and various devices and systems in emergency.

Note
You can change the pre-recorded message in InformaCast administration, as required.

Step 6 In the **Enter DN to trigger the panic button** field, enter the Directory Number (DN), which includes the digits 0 to 9, asterisks (*), and pound signs (#). Default value is ***5.

Step 7 From the **Route Partition** drop-down list, select a partition to restrict access to the route pattern.

Note
If you do not want to restrict access to the route pattern, select <None> for the partition.

Step 8 Click **Choose Phones to Send Notification** button.
The **Phones to Send Notification** dialog box appears.

Step 9 From the **Phones to Send Notification** dialog box, select the Cisco Unified IP phones to send the pre-recorded message. The dial pattern entered by you (for example, ***5) is configured as speed dial on the selected phones.
The selected Cisco Unified IP Phone are displayed in the **Selected Phones to Send Notification** list box.

Step 10 Click **Add Rules**, to create a new rule for the selected Cisco Unified IP Phone to receive notifications.

a) Select one of the parameters from the drop-down list. The available options are Device Pool, Description, and Directory Number.

b) In the second drop-down list, select a criteria from the following options:

- Does
- Does not

c) In the third drop-down list, select a criteria from the following options:

- Begins with
- Ends with
- Contains

d) In the text box, enter the search criterion.

Note
Minimum of one new rule and maximum of new five rules can be created. The **Add Rules** button gets disabled when five rules are created.

Note
To delete a rule, click **Delete Rules**.

e) Click **Test Rules**, to validate the created rules. When the test rule is completed with more than zero phones, the **Next** button is enabled.

Note
Phones added to Cisco Unified Communications Manager at a later date that match this rule will be included as recipients in notifications to this group.

Step 11 Click **Next**.

The wizard performs the following tasks:

- Adds a speed dial for the entered DN to the selected phones. If the selected phones have unused speed dials assigned to existing phone button templates, this speed dial appears directly on the selected phones. If the selected phones do not have unused speed dial buttons, the panic button speed dial is created, but it does not appear on the phone.
- Adds route pattern for entered DN in selected partition using created route list.
- Creates an InformaCast DialCast entry for the entered DN to send the selected message to the phones matching the selected rules.

Configure CallAware Emergency Call Alerting

Use this procedure to configure the CallAware emergency call alerting details. This sends a text and audio notification to IP phones when an emergency number is dialed. It can also detect calls to numbers other than 911.

Before you begin

Configure Panic Button, on page 307.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Advanced Features** > **Emergency Notifications Paging**.

Step 2 In the **Introduction to InformaCast Emergency Notifications** page, click **Next** to continue.

Step 3 In the **Installing the InformaCast Virtual Appliance** page, click **Next** to continue.

Step 4 In the **Connecting Cisco Unified Communications Manager and InformaCast** page, click **Next** to continue.

Step 5 In the **Configuring a Panic Button** page, click **Next** to continue.
The **Configuring CallAware Emergency Call Alerting** page appears.

Step 6 From the **Choose pre-recorded message by name** drop-down list, select the pre-recorded message to be displayed on Cisco Unified IP phones and various devices and systems in emergency.

Note
You can change the pre-recorded message in InformaCast administration, as required.

Step 7 Click **Choose Emergency Route Patterns** button.
The **Route Patterns** dialog box appears.

Step 8 From the **Route Patterns** dialog box, select the route patterns by checking the box next to the desired patterns.

a) Click the **Save Selected/Changes** button.

The selected route patterns are displayed in the **Selected Route Patterns** list box.

Step 9 Click **Add Rules**, to create a new rule for the selected Cisco Unified IP Phone to receive notifications.

a) Select one of the parameters from the drop-down list. The available options are Device Pool, Description, and Directory Number.

b) In the second drop-down list, select a criteria from the following options:

- Does
- Does not

c) In the third drop-down list, select a criteria from the following options:

- Begins with
- Ends with
- Contains

d) In the text box, enter the search criterion.

Note
Minimum of one new rule and maximum of five new rules can be created. The **Add Rules** button gets disabled when five rules are created.

Note
To delete a rule, click **Delete Rules**.

e) Click **Test Rules**, to validate the created rules. When the test rule is completed with more than zero phones, the **Finish** button is enabled.

Note

Phones added to Unified Communications Manager at a later date that match this rule will be included as recipients in notifications to this group.

Step 10 Click **Finish**.

The wizard performs the following tasks:

- Adds External Call Control profile for InformaCast
- For each selected route pattern, modify that route pattern to reference the External Call Control profile
- Creates a recipient group with rules that match phones to receive the notification
- Creates an InformaCast routing request with the selected message and recipient group

The **Summary** page appears and confirms the successful configuration of InformaCast with Unified Communications Manager. For more information, see https://www.singlewire.com.

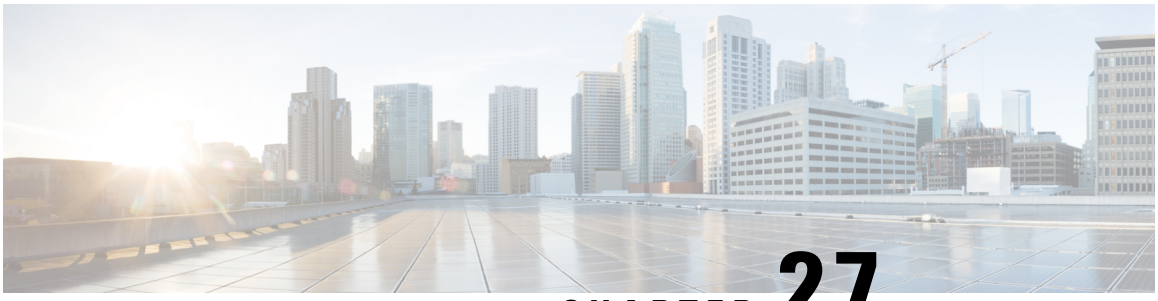
Paging Interactions

- Advanced Notification Paging Interactions, on page 310

Advanced Notification Paging Interactions

Table 27: Advanced Notification Paging Interactions

Feature	Interaction
Emergency Notifications Paging	You can configure the Emergency Notifications Paging wizard using InformaCast Release 11.5(1)SU3 and later versions in basic paging mode only. You can configure call monitoring to route patterns that contain digits only in the Emergency Notifications Paging wizard. For route patterns that contain wildcard characters, configure in InformaCast.



CHAPTER 27

Intercom

- Intercom Overview, on page 311
- Intercom Prerequisites, on page 312
- Intercom Configuration Task Flow, on page 312
- Intercom Interactions, on page 316
- Intercom Restrictions, on page 317
- Intercom Troubleshooting, on page 318

Intercom Overview

Intercom is a type of phone line that combines the functionality of a traditional line and a speed dial. With an intercom line, a user can call the intercom line of another user, which answers automatically to one-way audio whisper. The recipient can then acknowledge the whispered call and initiate a two-way intercom call.

You can use an intercom line to dial any other intercom line in the intercom partition, or you can preconfigure the line to target an intercom line outside the intercom partition.

Intercom allows a user to place a call to a predefined target. The called destination answers the call automatically in speakerphone mode with mute activated. This sets up a one-way voice path between the initiator and the destination, so the initiator can deliver a short message, regardless of whether the called party is busy or idle.

To ensure that the voice of the called party does not get sent back to the caller when the intercom call is automatically answered, Unified Communications Manager implements whisper intercom. Whisper intercom ensures that only one-way audio exists from the caller to the called party. The called party must manually press a key to talk to the caller.

An auto-answer tone indicates the beginning of the whisper intercom state for both the sender and the recipient.

Intercom and Default Devices

Each intercom line needs a default device. The intercom line is displayed only on the designated default device.

When the administrator assigns an intercom line to a device, the system sets the device as the default device for the intercom line if not set previously. The administrator can modify the default device for the intercom line. When the administrator changes the default device to a different device, the intercom line gets removed from the original device, even though the intercom line may still be assigned to the original device.

You can assign an intercom line to a device profile. Only when a user uses a device profile to log in to the default device that matches the default device of the intercom line does the intercom line become available. Otherwise, no intercom line is displayed when the user logs in.

Intercom Prerequisites

The intercom feature has the following system requirements:

- Cisco Unified IP Phones Firmware Release 8.3(1) or later

Intercom Configuration Task Flow

Before you begin

- Review Intercom Prerequisites, on page 312.

Procedure

	Command or Action	Purpose
Step 1	Configure Intercom Partition, on page 312	To add a new Intercom partition or configure an existing partition.
Step 2	Configure an Intercom Calling Search Space, on page 313	To add a new Intercom Calling Search Space.
Step 3	Configure an Intercom Translation Pattern, on page 314	To add a new Intercom Translation Pattern or to configure an existing Intercom Translation Pattern .
Step 4	Configure an Intercom Directory Number, on page 314	To add or update an Intercom Directory Number.
Step 5	Intercom Line and Speed Dial Configuration, on page 315	Configure Intercom Line and Speed Dial.

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 5

Procedure

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

The **Find and List Intercom Partitions** window appears.

Step 2 Click **Add New**.

An **Add New Intercom Partition** window appears.

Step 3 Under the **Intercom Partition Information** section, in the **Name** box, enter the name and description of the intercom partition that you want to add.

Note
To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (,) to separate the partition name and description on each line. If a description is not entered, Unified Communications Manager uses the partition name as the description.

Step 4 Click **Save**.

Step 5 Locate the partition that you want to configure.
Intercom Partition Configuration window is displayed

Step 6 Configure the fields in the Intercom Partition Configuration field area. See the online help for more information about the fields and their configuration options.

Step 7 Click **Save**.

The **Intercom Partition Configuration** window appears.

Step 8 Enter the appropriate settings. For detailed information about the Intercom Partition Configuration parameters, see online help.

Step 9 Click **Save**.

Step 10 Click **Apply Config**.

Configure an Intercom Calling Search Space

Before you begin

Configure Intercom Partition, on page 312

Procedure

Step 1 In the menu bar, choose **Call Routing** > **Intercom** > **Intercom Calling Search Space**.

Step 2 Click the **Add New**.

Step 3 Configure the fields in the Intercom Calling Search Space field area. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure an Intercom Translation Pattern

Before you begin

Configure an Intercom Calling Search Space, on page 313

Procedure

Step 1 Choose **Call Routing** > **Intercom** > **Intercom Translation Pattern**.

The **Find and List Intercom Translation Patterns** window appears.

Step 2 Perform one of the followings tasks:

a) To copy an existing intercom translation pattern, locate the partition to configure, click **Copy** eside the intercom translation pattern to copy.

b) To add a new intercom translation pattern, click the **Add New**.

Step 3 Configure the fields in the Intercom Translation Pattern Configuration field area. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Ensure that the intercom translation pattern that uses the selected partition, route filter, and numbering plan combination is unique. if you receive an error that indicates duplicate entries, check the route pattern or hunt pilot, translation pattern, directory number, call park number, call pickup number, or meet-me number configuration windows.

The **Intercom Translation Pattern Configuration** window displays the newly configured intercom translation pattern.

Configure an Intercom Directory Number

You can assign patterns to intercom directory numbers; for example, 352XX. To avoid user confusion, when you assign a pattern to an intercom directory number, add text or digits to these intercom DN configuration fields, Line Text Label, Display (Internal Caller ID), and External Phone Number Mask. These fields are displayed for an intercom directory number only after you add the intercom directory number and you associate the intercom directory number with a phone.

For example, add the username to the line text label and internal caller ID, and add the outside line number to the external number mask, when the calling information is displayed, it says John Chan, not 352XX.

Procedure

Step 1 Choose **Call Routing** > **Intercom** > **Intercom Directory Number**.

The **Find and List Intercom Directory Numbers** window is displayed.

Step 2 To locate a specific intercom directory number, enter search criteria and click **Find.**

A list of intercom directory numbers that match the search criteria displayed.

Step 3 Perform one of the followings tasks:

a) To add an intercom directory number, click **Add New**.

b) To update an intercom directory number, click the intercom directory number to update.

The **Intercom Directory Number Configuration** window displayed.

Step 4 Configure the fields in the Intercom Directory Number Configuration field area. For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save**.

Step 6 Click **Apply Config**.

Step 7 Click **Reset Phone**.

Step 8 Restart devices.

During the restart, the system may drop calls on gateways.

Intercom Line and Speed Dial Configuration

Before you begin

Configure an Intercom Directory Number, on page 314

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

Feature	Interaction
Bulk Administration Tool	The Unified Communications Manageradministrator can use the Bulk Administration Tool to add many intercom users at once instead of adding users individually. For more information, see Bulk Administration Guide for Cisco Unified Communications Manager.
Barge	When the intercom destination is a barge target, the Cisco Unified IP Phone can still support whisper intercom. 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.
Do Not Disturb (DND)	The intercom call will override DND on the destination phone.
Call Preservation	When a call is preserved, the end user must hang up before the phone can reregister with Unified Communications Manager. 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.)
Cisco Unified Survivable Remote Site Telephony (SRST)	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.
Cisco Unified Communications Manager Assistant	With the Unified Communications Manager Assistant Configuration Wizard, **Cisco Unified Communications Manager Assistant** configuration takes less time and eliminates errors. The partitions, calling search spaces, route point, and translation pattern automatically get created when the administrator successfully runs and completes the configuration wizard.
CTI	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.
Cisco Extension Mobility	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.

Feature	Interaction
Internet Protocol Version 6 (IPv6)	Intercom can support phones with an IP Addressing Mode of IPv4 Only or IPv4 and IPv6. During an intercom call, the talkback mode establishes media streams with the same IP version as the media stream that is used when the caller initiates intercom.
Intercom directory numbers (lines)	Intercom directory numbers (lines) are restricted to one device per intercom line. Cisco Extension Mobility is widely used; mobile users need the intercom feature but need it to be available only on a single device. You can assign intercom lines to either a regular device or to an extension mobility profile, but the system needs to enforce that an intercom line gets associated to either a regular device or to an extension mobility profile.
Extension mobility profile	An extension mobility profile can be used on more than one phone simultaneously, use the **Default Activated Device** field in the **Intercom Directory Number Configuration** window (**Cisco Unified CM Administration** > **Call Routing** > **Intercom** > **ntercom 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:

Feature	Restrictions
Hold	The system does not allow intercom calls to be placed on hold.
Call Forwarding	Intercom calls cannot be forwarded.
Transfer	The system does not allow an intercom call to be transferred.
iDivert	The system does not allow an intercom call to be diverted.
Call Pickup/Directed Call Pickup	The call pickup groups do not include intercom calls.
DND	Intercom overrides Do Not Disturb (DND).
Bandwidth	If sufficient bandwidth does not exist, the intercom call fails.
Call Target	If two intercom calls are directed to a target, the first one goes through; the second fails with a busy tone.
Barge and cBarge	Intercom does not work with Barge and cBarge.
Conferencing	The system does not allow intercom calls to be in conference.
Monitoring and Recording	When an active call is being monitored or recorded, the user cannot receive nor place intercom calls.

Feature	Restrictions
Video	Video is not supported with intercom.
Intercom Partition	An intercom partition assigned to an item such as calling search space or to a route pattern cannot be deleted.
Intercom Calling Search Spaces	Intercom calling search spaces that devices, lines (DNs), translation patterns, or other items are using cannot be deleted.

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.
- I

  f it is, ensure that the dialed-out DN is configured on another phone and that the phone is registered with the same 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 Unified Communications Manager Release 6.0(x) or later, includes two intercom lines. 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

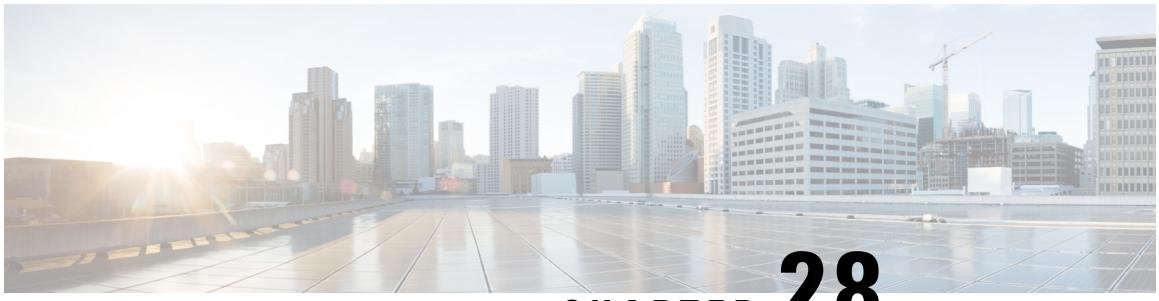
If the configuration has been done, reset the phone.



PART X

Receiving Calls

- Prime Line Support , on page 323
- Call Forwarding , on page 327
- Call Pickup , on page 353
- Call Park and Directed Call, on page 375
- Extension Mobility , on page 403
- Extension Mobility Cross Cluster , on page 421
- Extension Mobility Roaming Across Clusters, on page 455
- Hold Reversion , on page 469
- Accessing Hunt Groups , on page 477
- Malicious Call Identification , on page 485
- Call Transfer , on page 495
- External Call Transfer Restrictions , on page 509



CHAPTER 28

Prime Line Support

- Prime Line Support Overview, on page 323
- Prime Line Support Prerequisites, on page 323
- Prime Line Support Configuration Task Flow, on page 323
- Prime Line Support Interactions, on page 325
- Prime Line Support Troubleshooting, on page 326

Prime Line Support Overview

You can configure the Prime Line Support in Cisco Unified CM Administration so that when the phone is off-hook and receives a call on any line, the system always chooses the primary line for the call.

Prime Line Support Prerequisites

The following devices are compatible with the Prime Line Support feature:

Cisco Unified IP Phone 7900 Series, 8900 Series, and 9900 Series

For more information on the supported devices, see the latest version of *Cisco Unified IP Phone Guide* and *Cisco Unified IP Phone Administration Guide*.

Prime Line Support Configuration Task Flow

To configure the Prime Line Support feature for either the Cisco CallManager service or devices and device profiles, perform one of the following procedures.

Before you begin

- Review Prime Line Support Prerequisites, on page 323.

Procedure

	Command or Action	Purpose
Step 1	Configure Clusterwide Prime Line Support, on page 324	**(Optional)**. Configure the Prime Line Support feature for the Cisco CallManager service, which applies to the entire cluster.
Step 2	Configure Prime Line Support for Devices, on page 325	**(Optional)**. Configure the Prime Line Support feature for specific devices within the cluster, if you do not want to enable the feature clusterwide. **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 From Cisco Unified CM Administration, 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**- 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

Feature	Interaction
Always Use 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.
Maximum Number of Calls and Busy Trigger Settings	When the phone already has a call on a line, Unified Communications Manager uses the configuration for the Maximum Number of Calls and Busy Trigger settings to determine how to route the call.
Auto Answer	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.

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:

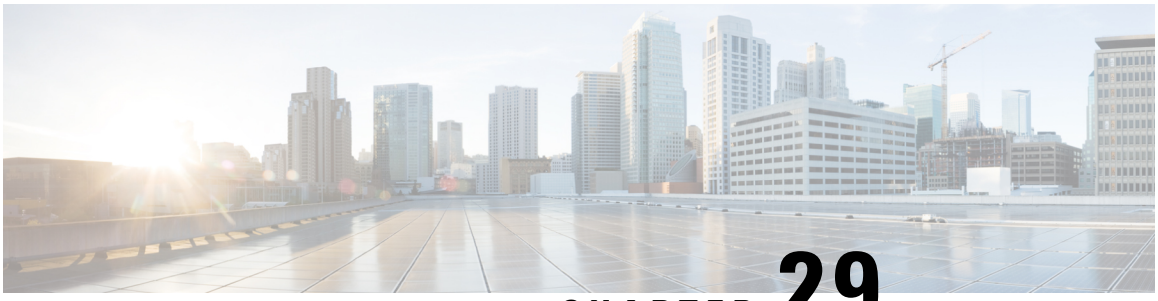
1. From Cisco Unified CM Administration, choose **System** > **Service Parameters**.
2. From the Server drop-down list, choose the server that is running the Cisco CallManager service.
3. From the Service drop-down list, choose **Cisco CallManager**.
4. In Cluster wide parameters (Device - phone), set **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. From 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**.



CHAPTER 29

Call Forwarding

- Call Forwarding Overview, on page 327
- Call Forwarding Configuration Task Flow, on page 329
- Call Forwarding Interactions, on page 346
- Call Forwarding Restrictions, on page 350

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). 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.

Unified Communications Manager prevents CFA activation on the phone when a CFA loop is identified. For example, 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, 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, Unified Communications Manager allows the CFA activation on the phone.

CFA loops do not affect call processing because 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. 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, Unified Communications Manager forwards the call to directory number 1004.

For a single call, 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

	Command or Action	Purpose
Step 1	Configure Partitions for Call Forwarding, on page 329	Administrators can configure partitions to restrict Call Forwarding to certain numbers based on the design criteria and requirements.
Step 2	Configure Calling Search Space for Call Forwarding, on page 331	Administrators can configure calling search spaces to restrict Call Forwarding to certain numbers based on the design criteria and requirements.
Step 3	Configure Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires, on page 332	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).
Step 4	Configure Call Forward No Bandwidth, on page 334	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.
Step 5	Configure Call Forward Alternate Destination, on page 335	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.
Step 6	Configure Other Call Forwarding Types, on page 336	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.
Step 7	Enable Destination Override for Call Forwarding, on page 346	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.

Configure Partitions for Call Forwarding

Configure partitions to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Partitions facilitate call routing by dividing the route plan into logical subsets that are based on organization, location, and call type. You can configure multiple partitions.

Configure partitions to restrict call forwarding to certain numbers based on your design criteria and requirements.

Procedure

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

Step 2 Click **Add New** to create a new partition.

Step 3 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 4 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 5 To create multiple partitions, use one line for each partition entry.

Step 6 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 7 Select one of the following radio buttons to configure the **Time Zone**:

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

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 28: Partition Name Guidelines

Partition Name Length	Maximum Number of Partitions
2 characters	340
3 characters	256
4 characters	204

Partition Name Length	Maximum Number of Partitions
5 characters	172
...	...
10 characters	92
15 characters	64

Configure Calling Search Space for Call Forwarding

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

Configure calling search spaces to restrict Call Forwarding to certain numbers based on your design criteria and requirements.

Before you begin

Configure Partitions for Call Forwarding, on page 329

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

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

Step 4 In the **Description** field, enter a description.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

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

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

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

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

Step 8 Click **Save**.

Configure Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires

The concept of hunting differs from that of call forwarding. Hunting allows Unified Communications Manager to extend a call to one or more lists of numbers, where each list specifies a hunting order that is chosen from a fixed set of algorithms. When a call extends to a hunt party from these lists and the party fails to answer or is busy, hunting resumes with the next hunt party. (The next hunt party varies depending on the current hunt algorithm.) Hunting then ignores the Call Forward No Answer (CFNA), Call Forward Busy (CFB), or Call Forward All (CFA) configured values for the attempted party.

Call Forwarding allows detailed control as to how to extend (divert or redirect) a call when a called party fails to answer, or is busy and hunting is not taking place. For example, if the CFNA value for a line is set to a hunt-pilot number, a call to that line that is not answered diverts to the hunt-pilot number and begins hunting.

Before you begin

Configure Calling Search Space for Call Forwarding, on page 331

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

Step 5 Click **Save**.

Hunt Call Treatment Fields for Call Forwarding

Field	Description
Hunt Call Treatment Settings **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.	

Field	Description

Forward Hunt No Answer	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:
	• **Do Not Forward Unanswered Calls**
	• **Use Forward Settings of Line Group Member** (replaces **Use Personal Preferences** check box)
	• **Forward Unanswered Calls to**
	  • **Destination**—Enter a directory number to which calls must be forwarded to.
	  • **Calling Search Space**—Choose a calling search space from the drop-down list which applies to all devices that use this directory number.
	• **Maximum Hunt Timer**—Enter a value (in seconds) that specifies the maximum time for hunting without queuing.
	Valid values are 1 to 3600. The default value is 1800 seconds (30 minutes).
	Caution
	Do not specify the same value for the Maximum Hunt Timer and the RNA Reversion Timeout on the associated line group.
	The forward no answer timer should be greater than the RNA timer of the line group.
	The forward no answer timer should not be multiples of RNA timer of line group.
	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.
	Note
	If hunting exceeds the number of hops that the **Forward Maximum Hop Count** service parameter specifies, hunting expires before the 30 minute maximum hunt timer value, and the caller receives a reorder tone.

Field	Description

Forward Hunt Busy	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:
	• **Do Not Forward Unanswered Calls**
	• **Use Forward Settings of Line Group Member**
	• **Forward Unanswered Calls to**
	  • **Destination**—Enter a directory number to which calls must be forwarded to.
	  • **Calling Search Space**—Choose a calling search space from the drop-down list which applies to all devices that use this directory number.

Configure Call Forward No Bandwidth

Before you begin

Configure Call Forwarding when Hunt List is Exhausted or Hunt Timer Expires, on page 332

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 call forward when there is insufficient bandwidth.
The **Directory Number Configuration** window is displayed.

Step 4 Configure the fields In the **AAR Settings** area. See Directory Number Configuration Fields for Call Forwarding, on page 335 for more information about the fields and their configuration options.

Step 5 Click **Save**.

Directory Number Configuration Fields for Call Forwarding

Field	Description
Voice Mail	Check this check box to forward the call to the voicemail. **Note** When you check this check box, Unified Communications Manager ignores the values in the **Destination** and **Calling Search Space** fields.
AAR Destination Mask	Enter a destination mask to determine the AAR destination to dial instead of using the external phone number mask.
AAR Group	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** > **Service Parameters**.
Retain this destination in the call forwarding history	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.

Configure Call Forward Alternate Destination

Before you begin

Configure Call Forward No Bandwidth, on page 334

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 MLPP Alternate Party And Confidential Access Level Settings Fields for Call Forwarding, on page 336 for more information about the fields and their configuration options.

Step 5 Click **Save**.

MLPP Alternate Party And Confidential Access Level Settings Fields for Call Forwarding

Field	Description
Target (Destination)	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 (*).
MLPP Calling Search Space	From the drop-down list, choose a calling search space to associate with the MLPP alternate party target (destination) number.
MLPP No Answer Ring Duration (seconds)	Enter the number of seconds (between 4 and 60) after which an MLPP precedence call will be directed to this directory number alternate party, if this directory number and its Call Forward destination have not answered the precedence call. You can also configure this value in the **Precedence Alternate Party Timeout** service parameter from **System** > **Service Parameters** from Cisco Unified CM Administration.

Configure Other Call Forwarding Types

You can configure Call Forward All (CFA), Call Forward Busy (CFB), Call Forward No Answer (CFNA), Call Forward No Coverage (CFNC), and Call Forward Unregistered (CFU) from the **Directory Number Configuration** window.

Before you begin

- For Call Forwarding functionality to work as intended, Cisco recommends that for the configured phones and the directory numbers in various partitions, the Call Forward Calling Search Spaces also be configured or else the forwarding may fail. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call.
- Configure Call Forward Alternate Destination, on page 335

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 Configure the **Call Forwarding and Call Pickup Settings** fields in the **Directory Number Configuration** window to configure CFA, CFB, CFNA, CFNC, and CFU. See Call Forwarding Fields, on page 337 for information about the fields and their configuration options.

Step 3 Click **Save**.

Call Forwarding Fields

Field	Description
Call Forward and Call Pickup Settings	

Field	Description
Calling Search Space Activation Policy	Three possible values exist for this option: • **Use System Default**—The **CFA CSS Activation Policy** service parameter determines which Forward All Calling Search Space to use for Call Forwarding. If the **CFA CSS Activation Policy** service parameter is set to **With Configured CSS**, then Forward All Calling Search Space and secondary Calling Search Space for Forward All will be used for Call Forwarding. This is the default setting. • **With Configured CSS**—The Forward All Calling Search Space that is explicitly configured in the **Directory Number Configuration** window controls the Forward All activation and Call Forwarding. If the Forward All Calling Search Space is set to **None**, no CSS is configured for Forward All. A Forward All activation attempt to any directory number with a partition will fail. No change in the Forward All Calling Search Space and secondary Calling Search Space for Forward All occurs during the Forward All activation. • **With Activating Device/Line CSS**—A combination of the Directory Number Calling Search Space and Device Calling Search Space controls the Forward All activation and Call Forwarding without explicitly configuring a Forward All Calling Search Space. When Forward All is activated from the phone, the Forward All Calling Search Space and secondary Calling Search Space for Forward All automatically gets populated with the Directory Number Calling Search Space and Device Calling Search Space for the activating device. If the Forward All Calling Search Space is set to **None**, and when Forward All is activated through the phone, the combination of Directory Number Calling Search Space and activating Device Calling Search Space controls the Forward All attempt. **CFA CSS Activation Policy**—Ensure that you configure this service parameter correctly for Forward All to work as intended in the **Service Parameter Configuration** window. The service parameter includes two possible values: • **With Configured CSS**—The primary and secondary CFA Calling Search Space controls the Call Forwarding attempt. • **With Activating Device/Line CSS**—The primary and secondary CFA Calling Search Space is updated with primary Line Calling Search Space and activating Device Calling Search Space. **Roaming**—When a device is roaming in the same device mobility group, Cisco Unified Communications Manager uses the Device Mobility CSS to reach the local gateway. If a user sets Call Forward All at the phone, the CFA CSS is set to **None**, and the CFA CSS Activation Policy is set to **With Activating Device/Line CSS**, then: • The Device CSS and Line CSS is used as the CFA CSS when the device is in its home location. • If the device is roaming within the same device mobility group, the Device Mobility CSS from the Roaming Device Pool and the Line CSS is used as the CFA CSS. • If the device is roaming within a different device mobility group, the Device CSS and Line CSS is used as the CFA CSS.

Field	Description
Forward All	The fields in this row of fields specify the Call Forwarding treatment for calls to this directory number if the directory number is set to forward all calls. The value in the **Calling Search Space** field is used to validate the Forward All destination that is entered when the user activates Call Forward All from the phone. This field is also used to redirect the call to the Call Forward All destination. Configure the following values: • **Voice Mail**—Check this check box to use the value that is set in the **Voice Mail Profile Configuration** window. **Note** When this check box is checked, Unified Communications Manager ignores the values in the **Destination** and **Calling Search Space** fields. • **Destination**—This field indicates the directory number to which all calls are forwarded. Use any dialable phone number, including an outside destination. • **Calling Search Space**—This value applies to all devices that use this directory number. • **Forward Maximum Hop Count**—Configure this parameter from the Cisco Unified CM Administrator, choose **System** > **Service Parameters**. This service parameter specifies the maximum number of times that a single call can be forwarded or diverted, and has special considerations for QSIG calls. For an incoming QSIG call, the maximum value is 15 (per ISO specifications); if you specify a greater value in this field, the specified value will apply to non-QSIG calls and for an incoming QSIG call, the call will only divert a maximum of 15 times. When QSIG trunks are configured, Cisco recommends setting this parameter to 15. For example, if the value of this parameter is seven, and a Call Forward All chain occurs consecutively from directory numbers 1000 to 007, which comprises seven hops, Cisco Unified Communications Manager prevents a phone user with directory number 2000 from activating CFA to directory number 1000, because no more than seven forwarding hops are supported for a single call.
Secondary Calling Search Space for Forward All	Because Call Forwarding is a line-based feature, in cases where the Device Calling Search Space is unknown, the system uses only the Line Calling Search Space to forward the call. If the Line Calling Search Space is restrictive and not routable, the forward attempt fails. Addition of a secondary calling search space for Call Forward All provides a solution to enable forwarding. The primary calling search space for Call Forward All and secondary calling search space for Call Forward All get concatenated (primary CFA CSS + secondary CFA CSS). Unified Communications Manager uses this combination to validate the CFA destination and to forward the call.

Field	Description
Forward Busy Internal	The fields in this row of fields specify the forwarding treatment for internal calls to this directory number if the directory number is busy. The values in the **Destination** and the **Calling Search Space** fields are used to redirect the call to the forward destination. Configure the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window for internal calls. **Note** When this check box is checked, the calling search space of the voicemail pilot is used. Unified Communications Manager ignores the values in the **Destination** and the **Calling Search Space** fields. **Note** When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the **Voice Mail** check box for external calls. • **Destination**—This field indicates the Call Forward Busy destination for internal calls. Use any dialable phone number, including an outside destination. **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls. • **Calling Search Space**—The Forward Busy Internal Calling Search Space is used to forward the call to the Forward Busy Internal destination. It applies to all devices that use this directory number. **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward Busy Calling Search Space. If you do not configure the Forward Busy Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails. **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls. The Call Forward Busy trigger is configured for each line appearance and cannot exceed the maximum number of calls that are configured for a line appearance. The Call Forward Busy trigger determines how many active calls exist on a line before the Call Forward Busy setting is activated (for example, ten calls). **Tip** Keep the busy trigger slightly lower than the maximum number of calls so that users can make outgoing calls and perform transfers. **Tip** If a call gets forwarded to a directory number that is busy, the call is not completed.

Field	Description
Forward Busy External	The fields in this row of fields specify the forwarding treatment for external calls to this directory number if the directory number is busy. The **Destination** and **Calling Search Space** fields is used to redirect the call to the forward destination. Configure the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window for external calls. **Note** When this check box is checked, the calling search space of the voicemail pilot is used. Unified Communications Manager ignores the values in the **Destination** and the **Calling Search Space** fields. **Note** When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the **Voice Mail** check box for external calls. • **Destination**—This field indicates the Call Forward Busy destination for external calls. Use any dialable phone number, including an outside destination. **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls. • **Calling Search Space**—The Forward Busy External Calling Search Space forwards the call to the Forward Busy External destination. It applies to all devices that use this directory number. **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward Busy Calling Search Space. If you do not configure the Forward Busy Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails. **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls.

Field	Description
Forward No Answer Internal	The fields in this row of fields specify the forwarding treatment for internal calls to this directory number if the directory number does not answer. The **Destination** and **Calling Search Space** fields are used to redirect the call to the forward destination. Configure the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window. **Note** When this check box is checked, the calling search space of the voicemail pilot is used. Unified Communications Manager ignores the values in the **Destination** and the **Calling Search Space** fields. **Note** When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the **Voice Mail** check box for external calls. • **Destination**—This field indicates the directory number to which an internal call is forwarded when the call is not answered. Use any dialable phone number, including an outside destination. **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls. • **Calling Search Space**—The Forward No Answer Internal Calling Search Space is used to forward the call to the Forward No Answer Internal destination. It applies to all devices that use this directory number. **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward No Answer destination, you should also configure the Forward No Answer Calling Search Space. If you do not configure the Forward No Answer Calling Search Space and the Forward No Answer destination is in a partition, the forward operation fails. **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls.

Field	Description
Forward No Answer External	The fields in this row of fields specify the forwarding treatment for external calls to this directory number if the directory number does not answer. The **Destination** and **Calling Search Space** fields are used to redirect the call to the forward destination. Configure the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window. **Note** When this check box is checked, the calling search space of the voicemail pilot is used. Unified Communications Manager ignores the values in the **Destination** and the **Calling Search Space** fields. **Note** When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to be forwarded to the voicemail system, you must uncheck the **Voice Mail** check box for external calls. • **Destination**—This field indicates the directory number to which an external call is forwarded when the call is not answered. Use any dialable phone number, including an outside destination. **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls. • **Calling Search Space**—The Forward No Answer External Calling Search Space is used to forward the call to the Forward No Answer External destination. It applies to all devices that use this directory number. **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to **None**, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward No Answer Calling Search Space. If you do not configure the Forward No Answer Calling Search Space and the Forward No Answer destination is in a partition, the forward operation fails. **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls.

Field	Description
Forward No Coverage Internal	The **Destination** and **Calling Search Space** fields are used to redirect the call to the forward destination. Configure the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window. **Note** When this check box is checked, Unified Communications Manager ignores the values in the **Destination** and **Calling Search Space** fields. When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to forward to the voicemail system, you must uncheck the **Voice Mail** check box for external calls. • **Destination**—This field specifies the directory number to which an internal nonconnected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination. **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls. • **Calling Search Space**—The Forward No Coverage Internal Calling Search Space is used to forward the call to the Forward No Coverage Internal destination. This value applies to all devices that use this directory number. **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured Call Forward Calling Search Space is used to forward the call. If the **Calling Search Space** field is set to None, the forward operation fails if the system uses partitions and calling search spaces. For example, if you configure the Forward Busy destination, you should also configure the Forward No Coverage Calling Search Space. If you do not configure the Forward No Coverage Calling Search Space and the Forward Busy destination is in a partition, the forward operation fails. **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, you must choose a different value in the **Calling Search Space** field for external calls.

Field	Description
Forward No Coverage External	The **Destination** and **Calling Search Space** fields are et used to redirect the call to the forward destination. Specify the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window. **Note** When this check box is checked, Unified Communications Manager ignores the values in the **Destination** and the **Calling Search Space** fields. When this check box is checked for internal calls, the system automatically checks the **Voice Mail** check box for external calls. If you do not want external calls to forward to the voicemail system, you must uncheck the **Voice Mail** check box for external calls. • **Destination**—This field specifies the directory number to which an internal nonconnected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination. **Note** When you enter a destination value for internal calls, the system automatically copies this value to the **Destination** field for external calls. If you want external calls to be forwarded to a different destination, you must enter a different value in the **Destination** field for external calls. • **Calling Search Space**—The Forward No Coverage External Calling Search Space is used to forward the call to the Forward No Coverage External destination. This value applies to all devices that use this directory number. **Note** If the system is using partitions and calling search spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces. When a call is forwarded or redirected to the Call Forward destination, the configured call forward calling search space is used to forward the call. If the **Calling Search Space** is **None**, the forward operation may fail if the system is using partitions and calling search spaces. For example, if you configure the Forward No Coverage destination, you should also configure the Forward No Coverage Calling Search Space. If you do not configure the Forward No Coverage Calling Search Space, and the Forward No Coverage destination is in a partition, the forward operation may fail. **Note** When you choose a calling search space for internal calls, the system automatically copies this value to the calling search space setting for external calls. If you want external calls to be forwarded to a different calling search space, choose a different value in the **Calling Search Space** field for external calls.
Forward on CTI Failure	This field applies only to CTI route points and CTI ports. The fields in this row specify the forwarding treatment for external calls to this CTI route point or CTI port if the CTI route point or CTI port fails. Configure the following values: • **Voice Mail**—Check this check box to use the configured values in the **Voice Mail Profile Configuration** window. **Note** When this check box is checked, Unified Communications Manager ignores the values in the **Destination** and **Calling Search Space** fields. • **Destination**—This field specifies the directory number to which an internal nonconnected call is forwarded when an application that controls that directory number fails. Use any dialable phone number, including an outside destination. • **Calling Search Space**—This value applies to all devices that use this directory number.
Forward Unregistered Internal	This field applies to unregistered internal DN calls. The calls are rerouted to a specified destination or voicemail. **Note** You must also specify the maximum number of forwards in the **Service Parameters Configuration** window for a directory number in the **Max Forward UnRegistered Hops to DN** service parameter. This parameter specifies the maximum number of forward unregistered hops that are allowed for a directory number at the same time. This parameter limits the number of times the call can be forwarded due to unregistered DN when a forwarding loop occurs. Use this count to stop forward loops for external calls that have been Call Forward Unregistered. Unified Communications Manager terminates the call when the value that is specified in this service parameter is exceeded.

Field	Description
Forward Unregistered External	This field applies to unregistered external DN calls. The calls are rerouted to a specified destination or voicemail. **Note** You must also specify the maximum number of forwards in the **Service Parameters Configuration** window for a directory number in the **Max Forward UnRegistered Hops to DN** service parameter. This parameter specifies the maximum number of forward unregistered hops that are allowed for a directory number at the same time. This parameter limits the number of times the call can be forwarded due to unregistered DN when a forwarding loop occurs. Use this count to stop forward loops for external calls that have been Call Forward Unregistered. Unified Communications Manager terminates the call when the value that is specified in this service parameter is exceeded.
No Answer Ring Duration (seconds)	This field specifies the seconds to wait before forwarding the unanswered call to the Call Forward No Answer destination, if specified. Make sure the value that is specified in this parameter is less than the value that is specified in the **T301 Timer** service parameter. If the value in the **Forward No Answer Timer** service parameter is greater than the value that is specified in the **T301 Timer** service parameter, the call is not forwarded and the caller receives a busy signal. Leave this field empty if you want to set the value in the Cisco Unified Communications Manager **Forward No Answer Timer** service parameter.

Enable Destination Override for Call Forwarding

Enable the destination override for call forwarding, Unified Communications Manager ignores the CFA destination when it matches the calling party number. The override applies to both internal and external calls.

In cases where the calling party number has been transformed, the calling party number does not match the CFA destination, no override occurs.

Before you begin

Configure Other Call Forwarding Types, on page 336

Procedure

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

Step 2 In the **Clusterwide Parameters (Feature - Hold Reversion)** area, set the **CFA Destination Override** service parameter value to **True**.

Call Forwarding Interactions

Feature	Interaction
Call Back	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.

Feature	Interaction
Call Display Restrictions	The Connected Number Display restriction applies to all calls that originate in the system. When this value is set to **True**, this field interacts transparently with existing 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.
Do Not Disturb	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.
External Call Control	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, 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 Unified Communications Manager. **Note** The **Call Diversion Hop Count** service parameter that supports External Call Control, and the **Call Forward Call Hop Count** service parameter that supports Call Forwarding are independent of each other; they work separately.
Extension Mobility Cross Cluster	Cisco Extension Mobility Cross Cluster supports Call Forwarding.
Extend and Connect	Extend and Connect supports Call Forward All.
Immediate Divert	When the Forward No Answer field in the **Directory Number Configuration** window is not configured, Call Forward uses the clusterwide CFNA timer service parameter, **Forward No Answer Timer**. If a user presses the iDivert softkey at the same time as the call is being forwarded, the call gets diverted to an assigned call forward directory number (because the amount of time set on the timer was too short), not the voicemail. To resolve this situation, set the CFNA timer service parameter to enough time (for example, 60 seconds).
Logical Partitioning	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.

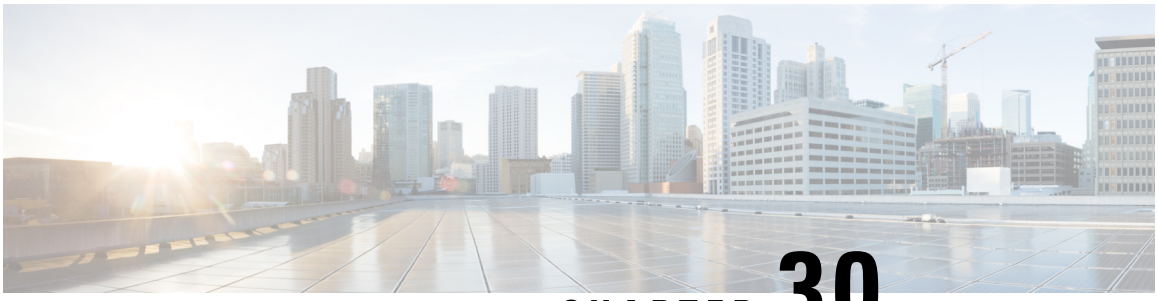
Feature	Interaction
Multilevel Precedence and Preemption (MLPP)	**Call Forward Busy** • You can optionally configure a preconfigured Precedence Alternate Party target for any MLPP-enabled station. • Cisco Unified Communications Manager applies the Call Forward Busy feature to forward a precedence call in the usual manner before it applies to any Precedence Alternate Party Diversion procedures to the call. • The system preserves precedence of calls across multiple forwarded calls. • If the incoming precedence call is of higher precedence than the existing call, preemption occurs. Both the preempted parties in the active call receive a continuous preemption tone until the station to which the precedence call is directed hangs up. After hanging up, the station to which the precedence call is directed receives precedence ringing. The destination station connects to the preempting call when the station goes off hook. **Call Forward No Answer** • For calls of Priority precedence level and above, call processing preserves the precedence level of calls during the forwarding process and may preempt the forwarded-to user. • If an Alternate Party is configured for the destination of a precedence call, call processing diverts the precedence call to the Alternate Party after the Precedence Call Alternate Party timeout expires. If no Alternate Party value is configured for the destination of a precedence call, call processing diverts the precedence call to the Call Forward No Answer value. • Normally, precedence calls are routed to users and not to the voicemail system. The administrator sets the **Use Standard VM Handling For Precedence Calls** enterprise parameter to avoid routing precedence calls to voicemail systems. If the incoming precedence call is of equal or lower precedence than the existing call, call processing invokes normal call-forwarding behavior. If the destination station for a precedence call is nonpreemptable (that is, not MLPP-configured), call processing invokes call-forwarding behavior. Alternate Party Diversion (APD) comprises a special type of call forwarding. If users are configured for APD, APD takes place when a precedence call is directed to a directory number (DN) that is busy or does not answer. MLPP APD applies only to precedence calls. An MLPP APD call disables the DN Call Forward No Answer value for precedence calls.

Feature	Interaction
Originally called party name in Placed Call History	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.
Rollover Lines	By using call forwarding settings, you can create rollover lines for a shared line. This could be a useful for some call center situations. With rollover lines, when someone dials a number (e.g. 1-800-HOTLINE), the call always is routed to a specific phone line. This may be a shared line that is shared by multiple phones. If line 1 is busy, the call rolls over to line 2, if line 2 is busy it rolls over to line 3, and so on. Line 2 or 3 become available only if line 1 is busy. This type of call functionality is possible via call forwarding busy settings and the Busy Trigger as follows: • On line 1, set the Busy Trigger to 1 and configure Call Forward Busy to the second line in the chain. • On line 2, set the Busy Trigger to 1 and configure Call Forward Busy to the third line in the chain • Continue this for as many lines as meets your needs.
Secure Tone	Call Forward All is supported on protected phones.
Session Handoff	When the user hands off a call, a new call gets presented on the desk phone. While the desk phone is flashing, Call Forward All is not triggered on the desk phone for the call that was handed off.
Shared Lines	When you use a shared line with Call Forward All (CFA) setting and choose the 'Calling Number' as 'Redirected Party's External Phone Number' presentation in the outgoing trunk, then the redirected number that is displayed may not be consistent when shared lines have different E164 numbers configured. So we recommend using the same E164 number across Shared lines.

Call Forwarding Restrictions

Feature	Restriction
Call Forwarding	• If Call Forward All activation occurs in Unified Communications Manager or the Cisco Unified Communications Self Care Portal, Unified Communications Manager does not prevent the CFA loop. • 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). 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, 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. • You cannot activate Call Back if you forward all calls to voicemail system. • 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.
Immediate Divert	When Call Forward All (CFA) and Call Forward Busy (CFB) are activated, the system does not support Immediate Divert (CFA and CFB have precedence over Immediate Divert).
Intercom	You cannot forward Intercom calls.
Log Out of Hunt Group	When a phone that is running SIP (7906, 7911, 7941, 7961) is logged in to hunt groups and Call Forward All is activated, the call gets presented to the phone that is running SIP. When 7940 and 7960 IP phones that are running SIP are logged in to hunt groups and Call Forward All is activated, the phone gets skipped and the next phone in the line group is rung.

Feature	Restriction
Logical Partitioning	Logical partitioning handling does not take place in the following circumstances: • When both the caller and forwarded devices are Voice over IP (VoIP) phones. • When geolocation or a geolocation filter is not associated with any device.
Multilevel Precedence and Preemption (MLPP)	Multilevel Precedence and Preemption (MLPP) support for supplementary services specifies the following restrictions for Call Forwarding: • Call Forward All (CFA) support for inbound MLPP calls always forwards the call to the MLPP Alternate Party (MAP) target of the called party, if the MAP target is configured. In the event of an incorrect configuration (that is, if no MAP target is specified), the call gets rejected, and the calling party receives reorder tone. • Call Forward No Answer (CFNA) support for inbound MLPP calls forwards the call once to a CFNA target. After the first hop, if the call remains unanswered, the call is sent to the MAP target of the original called party, if the MAP target is configured. In the event of an incorrect configuration (that is, if no MAP target is specified), the call gets rejected, and the calling party receives reorder tone. • Call Forward Busy (CFB) support for inbound MLPP calls forwards the call up to the maximum number that is configured for forwarding hops. If the maximum hop count is reached, the call is sent to the MAP target of the original called party, if the MAP target is configured. In the event of an incorrect configuration (that is, no MAP target is specified), the call gets rejected, and the calling party receives reorder tone.
Call Forward Classification with Call Transfer	When a call is transferred, the call classification takes on the classification of the transferred leg, rather than the original leg. For example: • Incoming call from PSTN is received by a receptionist. This is an external call. • The receptionist transfers the call to extension 3100. The transferred call is now an internal call. • The user at extension 3100 is busy, but has Call Forward External configured to send external calls back to the receptionist. However, because the call takes on the classification of the second leg (internal), the call goes to voicemail.



CHAPTER 30

Call Pickup

- Call Pickup Overview, on page 353
- Call Pickup Configuration Task Flow, on page 355
- Call Pickup Interactions, on page 373
- Call Pickup Restrictions, on page 373

Call Pickup Overview

The Call Pickup feature allows users to answer calls that come in on a directory number other than their own.

Consider a setup where phone A is the calling phone. Phone B and phone C are both associated with the same call pickup group. If phone A calls phone B, phone B rings and phone C get the notification. To answer the call on phone C, the user must press the **Pickup** softkey and answer the call to make the connection.

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. Users use the **GPickUp** softkey for this type of call pickup.

When a 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.

Consider a setup where phone A is the calling phone and there are two call pickup groups namely, X and Y. Call pickup group X is associated with phone B and phone C and call pickup group Y is associated with phone D. If phone A calls phone B, phone B rings and phone C gets the notification. But phone D does not get any notification as it belongs to a pickup group that is different from the one that phone B is associated with. To answer the call on phone D, the user must press the **GPickup** softkey, dial the group number of call pickup group X. Now phone D starts ringing and when the user answers the call on phone D, phone A and phone D are connected.

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 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 **OPickUp** softkey for this type of call pickup. If Auto Other Group Pickup is not enabled, the user must press the OPickUp and Answer softkeys 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 for answering calls. The group X picks up an incoming call in group A, though a call may have come in group C before the incoming call in group A.

Note 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.

Note

- Each call pickup group can be associated with a maximum of 10 other call pickup groups.
- Users can be assigned to only one call pickup group.
- Multiple call pickup groups can exist within the same location, but each group name must be unique within that location.

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. 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 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

	Command or Action	Purpose
Step 1	Configure a Call Pickup Group, on page 357	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 358	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.

	Command or Action	Purpose
		Repeat this procedure for each call pickup group that you create.
Step 3	Create another call pickup group and associate it with the BLF call pickup group that you created in Step 1, on page 355. You can associate a call pickup group with multiple BLF DN call pickup groups.	Perform this step if you are configuring BLF Call Pickup. **Note** You do not always need to create another call pickup group. For example, you can have a single call pickup group that includes both the initiator DN and the destination DN. In such cases, associate the BLF call pickup group with itself.
Step 4	Configure Partitions for Call Pickup, on page 358	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 5	Configure Calling Search Space, on page 359	If you configure partitions, you must also configure calling search spaces. Configure calling search spaces to identify the partitions that calling devices can search when they attempt to complete a call. You must complete this procedure for directed all pickup. It is optional for other types of call pickup.
Step 6	Assign a Call Pickup Group to Hunt Pilots, on page 360	**(Optional).** 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.
Step 7	Configure notifications: • Configure Call Pickup Notification, on page 361 • Configure Call Pickup Notification for a Directory Number, on page 362	**(Optional).** Configure notifications when other members of a pickup group receive a call. You can configure audio or visual notifications, or both.

	Command or Action	Purpose
	• Configure BLF Call Pickup Notification, on page 363	
Step 8	Configure Directed Call Pickup: • Configure a Time Period, on page 364 • Configure Time Schedule, on page 365 • Associate a Time Schedule with a Partition, on page 365	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 9	Configure automatic call answering: • Configure Auto Call Pickup, on page 366 • Configure BLF Auto Pickup, on page 367	**(Optional).** Enable automatic call answering and configure timers for automatic call answering.
Step 10	Configure phone button templates: • Configure Call Pickup Phone Button Template, on page 367 • Associate Call Pickup Button Template with Phone, on page 368 • Configure BLF Speed Dial Number for the BLF Call Pickup Initiator, on page 369	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 11	Configure Softkeys for Call Pickup, on page 369 • Configure a Softkey Template for Call Pickup, on page 370 • Associate a Softkey Template with a Common Device Configuration, on page 371 • Associate a Softkey Template with a Phone , on page 372	Configure softkeys for any of the call pickup features that you want to use: • Call Pickup (Pickup) • Group Call Pickup (GPickup) • Other Group Pickup (OPickup) For Directed Call Pickup, use the Group Call Pickup softkey.

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

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 357

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 358
- Configure Calling Search Space, on page 359

Configure Partitions for Call Pickup

You can restrict access to call pickup groups by assigning a partition to the call pickup group number. When this configuration is used, only the phones that have a calling search space that includes the partition with the

call pickup group number can participate in that call pickup group. Make sure that the combination of partition and group number is unique throughout the system. You can create multiple partitions.

If you assign call pickup group numbers to a partition, only those phones that can dial numbers in that partition can use the call pickup group. If partitions represent tenants in a multitenant configuration, make sure that you assign the pickup groups to the appropriate partition for each tenant.

Before you begin

Assign a Call Pickup Group to Directory Numbers, on page 358

Procedure

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

Step 2 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 3 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 4 To create multiple partitions, use one line for each partition entry.

Step 5 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 6 Select one of the following radio buttons to configure the **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 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 7 Click **Save**.

Configure Calling Search Space

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

Before you begin

Configure Partitions for Call Pickup, on page 358

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

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

Step 4 In the **Description** field, enter a description.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

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

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

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

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

Step 8 Click **Save**.

Assign a Call Pickup Group to Hunt Pilots

Only hunt lists that are assigned to a call pickup group can use Call Pickup, Group Call Pickup, BLF Call Pickup, Other Group Pickup, and Directed Call Pickup. Follow these steps to assign a call pickup group to hunt pilots:

Before you begin

Configure Calling Search Space, on page 359

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 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.

Before you begin

Assign a Call Pickup Group to Hunt Pilots, on page 360

Procedure

	Command or Action	Purpose
Step 1	Configure Call Pickup Notification for a Call Pickup Group, on page 361	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.
Step 2	Configure Call Pickup Notification for a Directory Number, on page 362	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 363	

Configure Call Pickup Notification for a Call Pickup Group

Before you begin

Assign a Call Pickup Group to Hunt Pilots, on page 360

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 Call Pickup Notification Fields for Call Pickup, on page 362 for details about the fields and their configuration options.

Note
Refer to **Call Pickup Interactions and Restrictions** for feature interactions and restrictions that will affect your Call Pickup configuration.

Call Pickup Notification Fields for Call Pickup

Field	Description
Call Pickup Group Notification Policy	From the drop-down list, select the notification policy. The available are options are No Alert, Audio Alert, Visual Alert, and Audio and Visual Alert.
Call Pickup Group Notification Timer	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.
Calling Party Information	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. **Note** The notification is sent only to the primary line of a device.
Called Party Information	Check the check box if you want the visual notification message to the call pickup group to include identification of the original called party. The system makes this setting available when the Call Pickup Group Notification Policy is set to Visual Alert or Audio and Visual Alert.

Configure Call Pickup Notification for a Directory Number

Perform these steps to configure the type of audio notification that is provided when a phone is idle or in use.

Before you begin

Configure Call Pickup Notification for a Call Pickup Group, on page 361

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 BLF Call Pickup Notification

Before you begin

Configure Call Pickup Notification for a Directory Number, on page 362

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 from **Clusterwide Parameters (Device - Phone** section in the **Service Parameter Configuration** window. See Service Parameter Fields for BLF Call Pickup Notification, on page 363 for more information about the fields and their configuration options.

Service Parameter Fields for BLF Call Pickup Notification

Field	Description
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

Field	Description
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
BLF Pickup Group Audio Alert Setting of Busy Station	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 Directed Call Pickup

Procedure

	Command or Action	Purpose
Step 1	Configure a Time Period, on page 364	Configure time period for members of the associated groups to your group.
Step 2	Configure Time Schedule, on page 365	Configure time schedule for members of the associated groups to your group.
Step 3	Associate a Time Schedule with a Partition, on page 365	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.

Configure a Time Period

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

Procedure

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

Step 2 Configure the fields in the **Time Period Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 3 Click **Save**.

Configure Time Schedule

Before you begin

Configure a Time Period, on page 364

Procedure

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

Step 2 Configure the fields in the **Time Schedule Configuration** window. For more information on the fields and their configuration options, see Online Help.

Associate a Time Schedule with a Partition

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

Before you begin

Configure Time Schedule, on page 365

Procedure

Step 1 From Cisco Unified CM Administration, 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

	Command or Action	Purpose
Step 1	Configure Auto Call Pickup, on page 366	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.
Step 2	Configure BLF Auto Pickup, on page 367	

Configure Auto Call Pickup

Auto call pickup connects the user to an incoming call. When the user presses the softkey on the phone, Unified Communications Manager locates the incoming call in the group and completes the call connection. You can automate call pickup, group pickup, other group pickup, directed call pickup, and BLF call pickup. If you do not enable automatic call answering, users must press additional softkeys or dial group numbers to complete the connection.

Before you begin

Associate a Time Schedule with a Partition, on page 365

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 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

Before you begin

Configure Auto Call Pickup, on page 366

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 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

	Command or Action	Purpose
Step 1	Configure Call Pickup Phone Button Template, on page 367	Add Call Pickup feature to the phone button template.
Step 2	Associate Call Pickup Button Template with Phone, on page 368	
Step 3	Configure BLF Speed Dial Number for the BLF Call Pickup Initiator, on page 369	

Configure Call Pickup Phone Button Template

Follow these steps to add Call Pickup feature to the phone button template.

Before you begin

Configure Automatic Call Answering, on page 366

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate Call Pickup Button Template with Phone

Before you begin

Configure Call Pickup Phone Button Template, on page 367

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure BLF Speed Dial Number for the BLF Call Pickup Initiator

Before you begin

Associate Call Pickup Button Template with Phone, on page 368

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

Note By default, the softkeys for Call Pickup are not available on a phone. You must configure a softkey template for any softkey to appear on the phone and be able to use it.

Procedure

	Command or Action	Purpose
Step 1	Configure a Softkey Template for Call Pickup, on page 370	Add the Pickup, GPickup, and OPickup softkeys to a softkey template.
Step 2	To Associate a Softkey Template with a Common Device Configuration, on page 371, complete the following subtasks: • Add a Softkey Template to Common Device Configuration, on page 372 • Associate a Common Device Configuration with a Phone, on page 372	**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 372	**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

	Command or Action	Purpose
		conjunction with the Common Device Configuration if you need assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey.

Configure a Softkey Template for Call Pickup

Use this procedure to make the following call pickup softkeys available:

Softkey	Description	Call States
Call Pickup (**Pickup**)	Allows you to answer a call on another extension in your group.	On Hook Off Hook
Group Call Pickup (**GPickup**)	Allows you to answer a call on extension outside your group.	On Hook Off Hook
Other Group Pickup (**OPickup**)	Allows you to answer a call ringing in another group that is associated with your group.	On Hook Off Hook

Before you begin

Configure Call Pickup Phone Buttons, on page 367

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one of the following tasks:

- Associate a Softkey Template with a Common Device Configuration, on page 371
- Associate a Softkey Template with a Phone , on page 372

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 372.

Procedure

Step 1 Add a Softkey Template to Common Device Configuration, on page 372

Step 2 Associate a Common Device Configuration with a Phone, on page 372

Add a Softkey Template to Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey Template with a Phone

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

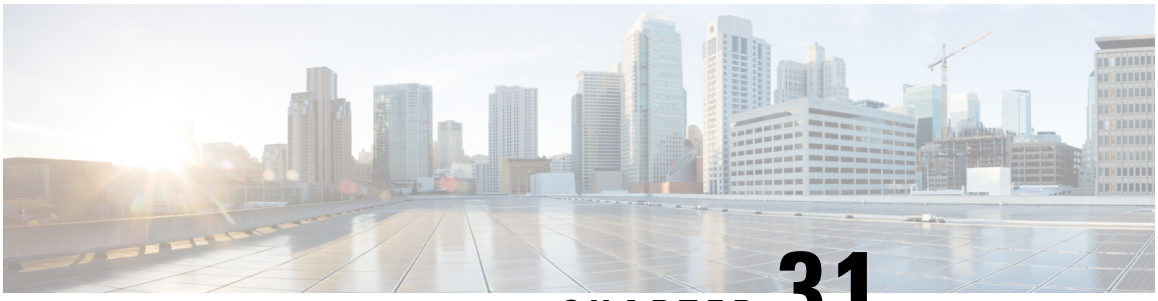
Call Pickup Interactions

Feature	Interaction
Route Plan Report	The route plan report displays the patterns and DNs that are configured in Unified Communications Manager. Use the route plan report to look for overlapping patterns and DNs before assigning a DN to call pickup group.
Calling search space and partitions	Assigning a partition to the Call Pickup Group number limits call pickup access to users on the basis of the device calling search space.
Time of Day (TOD)	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

Restriction	Description
Different phone lines to different call pickup groups	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.

Restriction	Description
Call Pickup Group Number	• You cannot delete a call pickup group number when it is assigned to a line or DN. To determine which lines are using the call pickup group number, use Dependency Records in **Call Pickup Configuration** window. To delete a call pickup group number, reassign a new call pickup group number to each line or DN. • When you update a call pickup group number, Cisco Unified Communications Manager automatically updates all directory numbers that are assigned to that call pickup group.
SIP Phones	• The system does not support Call Pickup Notification on a few Cisco Unified IP Phones that run SIP. • Call Pickup Notification is only supported on licensed, third-party phones that run SIP.
Directed Call Pickup	• If a device that belongs to a hunt list rings due to a call that was made by calling the hunt pilot number, users cannot use the Directed Call Pickup feature to pick up such a call. • Users cannot pick up calls to a DN that belongs to a line group by using the Directed Call Pickup feature.
BLF Pickup	The system does not support Call Pickup Notification on a few Cisco Unified IP Phones that run SIP.
Incoming Calling Party International Number Prefix - Phone	If you have configured a prefix in the “Incoming Calling Party International Number Prefix - Phone ” service parameter, and an international call is placed to a member in the Call Pickup Group, the prefix does not get invoked in the calling party field if the call gets picked up by another member of the Call Pickup Group.



CHAPTER 31

Call Park and Directed Call

- Call Park Overview, on page 375
- Call Park Prerequisites, on page 376
- Call Park Configuration Task Flow, on page 377
- Call Park Interactions, on page 391
- Call Park Restrictions, on page 392
- Troubleshooting Call Park, on page 392
- Directed Call Park Overview, on page 393
- Directed Call Park Prerequisites, on page 393
- Directed Call Park Configuration Task Flow, on page 393
- Directed Call Park Interactions, on page 397
- Directed Call Park Restrictions, on page 399
- Troubleshooting Directed Call Park, on page 399

Call Park Overview

The Call Park feature allows you to place a call on hold so that it can be retrieved from another phone in the 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 Unified Communications Manager cluster, and each 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. If you use the same park ranges on different partitions, ensure that the users have only one partition in their CSS to be able to park and retrieve the calls. Having multiple partitions may lead to incorrect partition selection.

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 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, the Unified Communications Manager chooses the next Call Park extension number that is available and displays that number on the phone.

Park Monitoring

Park Monitoring is an optional Call Park feature where Cisco Unified Communications Manager monitors the status of a parked call until a timer expires. After the timer expires, the call is forwarded to a preassigned number, sent to voicemail, or returned to the call parker. You can apply park monitoring to phone lines and to hunt pilots.

Call Park Prerequisites

If you are using call park across clusters, you must have partitions and calling search spaces configured.

Table 29: Cisco Unified IP Phones That Support Park Softkey Template and Call Park Button Template

Phone Model	Supported in Softkey Template	Supported in Phone Button Template
Cisco Unified IP Phones 6900 series (except 6901 and 6911)	X	X
Cisco IP Phone 7800 Series	X	X
Cisco Unified IP Phones 7900 series (except 7921, 7925, 7936, 7937)	X	
Cisco IP Phone 8800 Series	X	X
Cisco Unified IP Phones 8900 series	X	X
Cisco Unified IP Phones 9900 series	X	X
Cisco Unified IP Phones 7900 series (except 7906, 7911, 7921, 7925, 7936, 7937)		X
Cisco Unified IP Phone 9800 series	X	X

Note You can configure Call Park on any line (except line 1) or button by using the programmable line key feature.

Call Park Configuration Task Flow

Before you begin

- Review Call Park Prerequisites, on page 376

Procedure

	Command or Action	Purpose
Step 1	Configure Clusterwide Call Park, on page 378	**(Optional).** Configure Call Park for the entire cluster, or use the procedure in step 3 to configure Call Park on servers within the cluster
Step 2	Configure a Partition for Call Park, on page 378	Create a partition to add a Call Park Number
Step 3	Configure a Call Park Number, on page 379	Configure a Call Park Number to use Call Park across servers in a cluster.
Step 4	Configure a Softkey Template for Call Park, on page 382	Add the Park softkey to a softkey template.
Step 5	To Associate a Softkey Template with a Common Device Configuration , on page 383, complete the following subtasks: • Add a Softkey Template to a Common Device Configuration , on page 383 • Associate a Common Device Configuration with a Phone, on page 384	**Optional.** To make the softkey template available to phones, you must complete either this step or the following step. Follow this step if your system uses a **Common Device Configuration** to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.
Step 6	Associate a Softkey with a Phone, on page 384	**Optional.** Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in conjunction with the Common Device Configuration. Use this procedure in conjunction with the Common Device Configuration if you need assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.
Step 7	To Configure Call Park Button, on page 385, complete the following subtasks: • Configure a Phone Button Template for Call Park, on page 385 • Associate a Button Template with a Phone, on page 385	

	Command or Action	Purpose
Step 8	Configure Park Monitoring, on page 386	Complete this optional task flow to add Park Monitoring to your Call Park configuration.

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**.

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 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 Unified Communications Manager and CTI Manager services.

Configure a Partition for Call Park

Configure partitions to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Partitions facilitate call routing by dividing the route plan into logical subsets that are based on organization, location, and call type. You can configure multiple partitions.

Before you begin

(Optional) Configure Clusterwide Call Park, on page 378

Procedure

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

Step 2 Click **Add New** to create a new partition.

Step 3 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 4 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 5 To create multiple partitions, use one line for each partition entry.

Step 6 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 7 Select one of the following radio buttons to configure the **Time Zone**:

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

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 Unified Communications Manager. Each 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 Unified Communications Manager Dialed Number Analyzer tool.

Before you begin

Configure a Partition for Call Park, on page 378

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 Call Park Configuration Fields, on page 381 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.**

Call Park Configuration Fields

Field	Description
Call Park Number/Range	Enter the Call Park extension number. You can enter digits or the wildcard character X (the system allows one or two Xs). For example, enter 5555 to define a single Call Park extension number of 5555 or enter 55XX to define a range of Call Park extension numbers from 5500 to 5599. **Note** You can create a maximum of 100 Call Park numbers with one call park range definition. Make sure that the call park numbers are unique. **Note** You cannot overlap call park numbers between Unified Communications Manager servers. Ensure that each Unified Communications Manager server has its own number range. **Note** The call park range is selected from the list of servers where the call originates. For example, if phoneA (registered to nodeA) calls phone B (registered to nodeB) and the phoneB user presses Park, phoneB requires a call park range in the CSS that resides on nodeA. In a multinode environment where phones and gateways communicate with various nodes and where calls that originate from any server may need to be parked, the phones require a CSS that contains call park ranges from all servers.
Description	Provide a brief description of this call park number. The description can include up to 50 characters in any language, but it cannot include double-quotes (“), percentage sign (%), ampersand (&), or angle brackets (<>).
Partition	If you want to use a partition to restrict access to the call park numbers, choose the desired partition from the drop-down list. If you do not want to restrict access to the call park numbers, choose <None> for the partition. **Note** Make sure that the combination of call park extension number and partition is unique within the Unified Communications Manager.

Field	Description
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**.

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

Associate a Softkey Template with a Common Device Configuration

Optional. There are two ways to associate a softkey template with a phone:

- Add the softkey template to the **Phone Configuration**.
- Add the softkey template to the **Common Device Configuration**.

The procedures in this section describe how to associate the softkey template with a **Common Device Configuration**. Follow these procedures if your system uses a **Common Device Configuration** to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.

To use the alternative method, see the section *Associate a Softkey Template with a Phone* .

Procedure

Step 1 Add a Softkey Template to a Common Device Configuration , on page 383

Step 2 Associate a Common Device Configuration with a Phone, on page 384

Add a Softkey Template to a Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Configure Call Park Button

Configure a Phone Button Template for Call Park

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate a Button Template with a Phone

Before you begin

Configure a Phone Button Template for Call Park, on page 385

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure Park Monitoring

Complete these optional tasks to add Park Monitoring to your Call Park configuration.

Before you begin

Park Monitoring is supported on only a subset of phones that support Call Park. The following Cisco Unified IP Phones support Park Monitoring:

- Cisco IP Phone 8811
- Cisco IP Phone 8841
- Cisco IP Phone 8845
- Cisco IP Phone 8851
- Cisco IP Phone 8851NR
- Cisco IP Phone 8861
- Cisco IP Phone 8865
- Cisco IP Phone 8865NR
- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Procedure

	Command or Action	Purpose
Step 1	Configure Park Monitoring System Timers, on page 386	Configure system-level timers for the Park Monitoring feature.
Step 2	Configure Park Monitoring for Hunt Pilots, on page 387	**Optional.** If you have hunt pilots deployed, assign a Park Monitoring destination to a hunt pilot.
Step 3	Configure Park Monitoring for a Directory Number, on page 388	Assign a Park Monitoring destination for an individual phone line.
Step 4	Configure Park Monitoring via Universal Line Template, on page 389	If you have an LDAP directory sync configured, you can use universal line templates to provision directory number settings for multiple users with park monitoring configured.

Configure Park Monitoring System Timers

Use this procedure to configure system-level timers for the Park Monitoring feature.

Procedure

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

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

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

Step 4 Configure values for the following service parameters:

- **Park Monitoring Reversion Timer**—The number of seconds that Cisco Unified Communications Manager waits before prompting the user to retrieve a parked call. For individual phone lines, this setting can be overridden by the same setting in the **Directory Number Configuration** window. When the call park reversion timer expires, the call will be forwarded to the hunt pilot.
- **Park Monitoring Periodic Reversion Timer**—The number of seconds between reversion attempts when a call has been parked. Cisco Unified Communications Manager prompts the user about the parked call by ringing, beeping, or flashing the parker's phone. When the park monitoring reversion timer expires, the call will be forwarded to the parked party and not the hunt pilot.
- **Park Monitoring Forward No Retrieve Timer**—The number of seconds that park reminder notifications occur before the parked call is forwarded to the **Park Monitoring Forward No Retrieve** destination specified in the call parker's Directory Number configuration. When park monitoring forward no retrieve timer expires, the call will be forwarded to the hunt pilot.

Note
For additional details on these fields, see the service parameter online help.

Step 5 Click **Save**.

What to do next

Use any of these optional tasks to assign how expired timers get handled for individual phones lines and hunt pilots:

- Configure Park Monitoring for Hunt Pilots, on page 387
- Configure Park Monitoring for a Directory Number, on page 388
- Configure Park Monitoring via Universal Line Template, on page 389

Configure Park Monitoring for Hunt Pilots

If your deployment uses hunt pilots, use this optional procedure to assign a Park Monitoring destination to a hunt pilot.

Note For general information on setting up hunt pilots, see the "Configure Hunt Pilots" chapter of the System Configuration Guide for Cisco Unified Communications Manager.

Before you begin

Configure Park Monitoring System Timers, on page 386

Procedure

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

Step 2 Click **Find** and select the hunt pilot on which you want to configure a Park Monitoring destination.

Step 3 In the **Park Monitoring No Retrieve Destination** field, assign a **Destination** directory number and **Calling Search Space**.

Step 4 Complete any remaining fields in the **Hunt Pilot Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save.**

Configure Park Monitoring for a Directory Number

Use this procedure to assign a Park Monitoring destination for an individual phone line. You can forward calls to another number, send to voicemail, or return to the call parker.

Note The following tools are available to provision settings for multiple phone lines:

- Use a universal line template to provision park monitoring settings for multiple phone lines via an LDAP directory sync. For details, see Configure Park Monitoring via Universal Line Template, on page 389.
- Use the Bulk Administration Tool to import a CSV file with settings for a large number of phone lines. For more information, see the Bulk Administration Guide for Cisco Unified Communications Manager.

Before you begin

Configure Park Monitoring System Timers, on page 386

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Directory Number**.

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

Step 3 Enter values for the following **Park Monitoring** fields:

- **Park Monitoring Forward No Retrieve Destination External**—When the Park Monitoring Forward No Retrieve Timer expires, and the parkee is an external party, the call is forwarded either to voicemail or to a specified directory number. If this field s empty, the call is redirected to the call parker's line.
- **Park Monitoring Forward No Retrieve Destination Internal**—When the Park Monitoring Forward No Retrieve Timer expires, and the parkee is an internal party, the call is forwarded either to voicemail or to a specified directory number. If this field s empty, the call is redirected to the call parker's line.

- **Park Monitor Reversion Timer**—The number of seconds that Cisco Unified Communications Manager waits before prompting the user to retrieve a call parked on this phone line. If the value is 0 or empty, then Cisco Unified Communications Manager uses the value of the **Park Monitor Reversion Timer** service parameter.

Step 4 Complete any remaining fields in the **Directory Number Configuration** window.For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save**.

Configure Park Monitoring via Universal Line Template

Use this procedure to assign park monitoring settings to a universal line template. If you have an LDAP directory sync configured, you can use the universal line template configuration to provision directory number settings with park monitoring configured for multiple users.

Before you begin

Configure Park Monitoring System Timers, on page 386

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **User Phone/Add** > **Universal Line Template**.

Step 2 Perform one of the following steps:

- Click **Find** and select an existing template.
- Click **Add New** to create a new template.

Step 3 Expand the **Park Monitoring Settings** section and complete the fields. For field descriptions, see Park Monitoring Settings for Universal Line Templates, on page 389.

Step 4 Click **Save**.

What to do next

To apply the universal line template to individual directory numbers, you must assign the template to a user profile, feature group template, and LDAP directory sync. When the sync occurs, the template settings get applied to the phone lines that are a part of the sync. For LDAP setup, see the "Configure End Users" chapters in the System Configuration Guide for Cisco Unified Communications Manager.

Park Monitoring Settings for Universal Line Templates

The following table contains the Park Monitoring fields in the **Universal Line Template Configuration** window of Cisco Unified Communications Manager.

Table 30: Park Monitoring Settings for Universal Line Templates

Field	Description
Forward Destination for External Calls When Not Retrieved	When the person whose call is parked is an external party and the **Park Monitoring Forward No Retrieve Timer** expires, the system sends the call to one of these destinations: • **Voicemail**—Uses the configuration in Voice Mail Profile to determine where to send the call. • **Revert to Originator**—Returns the call to the call parker. • To forward calls to another number, input the other number in the text box. If no option is selected, the call returns to the call parker.
Calling Search Space for Forwarding External Calls When Not Retrieved	If you have configured parked calls to be redirected to a configured number, select the calling search space for the forward destination.
Forward Destination for Internal Calls When Not Retrieved	When the person whose call is parked is an internal party and the **Park Monitoring Forward No Retrieve Timer** expires, the system sends the call to one of these destinations: • **Voicemail**—Uses the configuration in Voice Mail Profile to determine where to send the call. • **Revert to Originator**—Returns the call to the call parker. • To forward calls to another number, input the other number in the text box. If no option is selected, the call returns to the call parker.
Calling Search Space for Forwarding Internal Calls When Not Retrieved	If you have configured parked calls to be redirected to a configured number, select the calling search space for the forward destination.
Park Monitor Reversion Timer (seconds)	This timer determines the number of seconds that Unified Communications Manager waits before prompting the user to retrieve a call that the user parked. This timer starts when the user presses the Park softkey on the phone, and a reminder is issued when the timer expires. The default value is 60 seconds. **Note** If you select 0 for the timer then phone lines that use this template will use the value of the **Park Monitor Reversion Timer** cluster-wide service parameter.

Call Park Interactions

Feature	Interaction
CTI Applications	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.
Music On Hold	Music On Hold allows users to place calls on hold with music that a streaming source provides. The Music On Hold audio source for Call Park is selected by the setting of the **Network Hold MOH Audio Source** setting within the **Phone Configuration** window. If you do not choose an audio source within the device configuration, Cisco Unified CM uses the audio source that is defined in the device pool or the system default if the device pool does not specify an audio source ID.
Route Plan Report	The route plan report displays the patterns and directory numbers that are configured in Unified Communications Manager. Use the route plan report to look for overlapping patterns and directory numbers before assigning a directory number to Call Park.
Calling Search Space and Partitions	Assign the call park directory number or range to a partition to limit call park access to users on the basis of the device calling search space.
Immediate Divert	Call Park supports Immediate Divert (iDivert or Divert softkey). For example, user A calls user B, and user B parks the call. User B retrieves the call and then decides to send the call to a voice-messaging mailbox by pressing the iDivert or Divert softkey. User A receives the voice mail greeting of user B.
Barge	• Barge with Call Park–The target phone (the phone that is being barged upon) controls the call. The barge initiator "piggybacks" on the target phone. The target phone includes most of the common features, even when the target is being barged; therefore, the barge initiator has no feature access. When the target parks a call, the barge initiator then must release its call (the barge). • cBarge with Call Park–The target and barge initiator act as peers. The cBarge feature uses a conference bridge, which causes it to function like a MeetMe conference. Both phones (target and barge initiator) have full access to their features.
Directed Call Park	We recommend that you do not configure both Directed Call Park and the Park softkey for Call Park, but the possibility exists to configure both. If you configure both, ensure that the call park and directed call park numbers do not overlap.
QSIG Intercluster Trunks	When a user parks a call across a QSIG intercluster trunk or a QSIG gateway trunk, the caller who has been parked (the parkee) does not see the To parked number message. The phone continues to display the original connected number. The call has been parked, and the user who parked the call can retrieve it. When the call is retrieved from the parked state, the call continues, but the caller who was parked does not see the newly connected number.

Call Park Restrictions

Feature	Restriction
Call Park	Unified Communications Manager can park only one call at each call park extension number.
Shared Line	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.
Backup	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.
Directed Call Park	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).
Conference	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, 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 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.

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 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 Configure Clusterwide Call Park, on page 378 for more information about the Timer.

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.

Unified Communications Manager can park only one call at each directed call park number. To retrieve a parked call, a user must dial a configured retrieval prefix followed by the directed call park number at which the call is parked.

Directed Call Park Prerequisites

Make sure that the phones in your deployment support Directed Call Park. For a list of supported phones, run the **Phone Feature List** report from Cisco Unified Reporting, selecting **Assisted Directed Call Park** as the feature. For details, see Generate a Phone Feature List, on page 5.

Directed Call Park Configuration Task Flow

Before you begin

- Review Directed Call Park Prerequisites, on page 393

Procedure

	Command or Action	Purpose
Step 1	Configure ClusterWide Directed Call Park, on page 394	To configure clusterwide parameter for directed call park.

	Command or Action	Purpose
Step 2	Configure a Directed Call Park Number, on page 394	To add, copy, and update a single Directed Call Park extension number or range of extension numbers.
Step 3	Configure BLF/Directed Call Park Buttons, on page 396	Configure a phone button template for BLF/Directed Call Park.
Step 4	Synchronize Directed Call Park with Affected Devices, on page 397	Synchronize Directed Call Park with Affected Devices

Configure ClusterWide Directed Call Park

Procedure

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

Step 2 To set the timer, update the **Call Park Reversion Timer** fields in the Clusterwide Parameter(Feature- General) section.

The default is 60 seconds. This parameter determines the time that a call remains parked. When this timer expires, the parked call returns to the device that parked the call or to another specified number, depending on what you configure in the **Directed Call Park Configuration** window.

Configure a Directed Call Park Number

Before you begin

Ensure that each directed call park directory number, partition, and range is unique within the Unified Communications Manager. Before you begin, generate a route plan report. If the Park softkey is also activated (not recommended), ensure that no overlap exists between call park numbers and directed call park numbers. If reversion number is not configured, the call reverts to the parker (parking party) after the Call Park Reversion Timer expires.

Configure ClusterWide Directed Call Park, on page 394

Procedure

Step 1 Choose **Call Routing** > **Directed Call Park**.

Step 2 Perform one of the following tasks:

- To add a new directed call park number, click **Add New**.
- To copy a directed call park number, find the directed call park number or range of numbers and then click the **Copy** icon.

- To update a directed call park number, find the directed call park number or range of numbers.

The **directed call park number configuration** window is displayed.

Step 3 Configure the fields in the Directed Call Park settings area. See Directed Call Park Configuration Settings, on page 395 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, 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** on the **Directed Call Park Configuration** window. This step is optional if you are using change notification.

Directed Call Park Configuration Settings

Field	Description
Number	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.
Description	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 (“), percentage sign (%), ampersand (&), or angle brackets (<>) and tabs.
Partition	If you want to use a partition to restrict access to the directed call park numbers, choose the desired partition from the drop-down list. If you do not want to restrict access to the directed call park numbers, leave the partition as the default of <None>. **Note** Make sure that the combination of directed call park number and partition is unique within Unified Communications Manager.
Reversion Number	Enter the number to which you want the parked call to return if not retrieved, or leave the field blank. **Note** A reversion number can comprise digits only; you cannot use wildcards.
Reversion Calling Search Space	Using the drop-down list, choose the calling search space or leave the calling search space as the default of <None>.

Field	Description
Retrieval Prefix	For this required field, enter the prefix for retrieving a parked call. The system needs the retrieval prefix to distinguish between an attempt to retrieve a parked call and an attempt to initiate a directed park.

Configure BLF/Directed Call Park Buttons

Before you begin

Configure ClusterWide Directed Call Park, on page 394

Procedure

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

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 BLF/Directed Call Park Configuration Fields, on page 396 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.

BLF/Directed Call Park Configuration Fields

Table 31: BLF/Directed Call Park Button Configuration Fields

Field	Description
Directory Number	The Directory Number drop-down list displays a list of Directed Call Park number that exist in the 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.
Label	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.

Field	Description
Label ASCII	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. **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.

Synchronize Directed Call Park with Affected Devices

Procedure

Step 1 Choose **Call Routing** > **Directed Call Park**.

The **Find and List Directed Call Parks** window is displayed.

Step 2 Choose the search criteria to use.

Step 3 Click **Find.**

The window displays a list of directed call parks that match the search criteria.

Step 4 Click the directed call park to which you want to synchronize applicable devices. The **Directed Call Park Configuration** window is displayed.

Step 5 Make any additional configuration changes.

Step 6 Click **Save.**

Step 7 Click **Apply Config**.

The **Apply Configuration Information** dialog is displayed.

Step 8 Click **OK.**

Directed Call Park Interactions

The following table describes feature interactions with the Directed Call Park feature.

Feature	Interaction
Music On Hold	The Music On Hold Audio Source for directed call park is assigned via the **Default Network Hold MOH Audio Source** service parameter. To assign the parameter: **1.** From Cisco Unified CM Administration, choose **System** > **Service Parameters**. **2.** From the **Server** drop-down list, choose a Unified Communications Manger cluster node. **3.** From the **Service** drop-down list, select **Cisco CallManager**. **4.** Under **Clusterwide Paramters (Service)**, assign a MOH audio source to the **Default Network Hold MOH Audio Source ID** parameter. The default is **1**. **5.** Click **Save**. **Note** For detailed information on adding MOH audio sources to the system, refer to the "Configure Music On Hold" section of this guide.
Calling Search Space and Partitions	Assign the Directed Call Park Directory number or range to a partition to limit Directed Call Park access to users on the basis of the device calling search space.
Immediate Divert	Directed call park supports Immediate Divert (iDivert or Divert softkey). For example, user A calls user B, and user B parks the call. User B retrieves the call and then decides to send the call to a voice-messaging mailbox by pressing the iDivert or Divert softkey. User A receives the voicemail greeting of user B.
Barge	• Barge with Directed Call Park–The target phone (the phone that is being barged upon) controls the call. The barge initiator "piggybacks" on the target phone. The target phone includes most of the common features, even when the target is being barged; therefore, the barge initiator has no feature access. When the target parks a call by using directed call park, the barge initiator then must release its call (the barge). • cBarge with Directed Call Park–The target and barge initiator act as peers. The cBarge feature uses a conference bridge that makes it behave like 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

Feature	Restriction
Directed Call Park number	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.
Standard Call Park Feature	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 Feature	We recommend that you do not press both the Transfer and Directed Call Park buttons simultaneously, as this may result in both DPark and Transfer failures.
Directed Call Park BLF	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.
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

User cannot retrieve parked calls. After dialing the directed call park number to retrieve a parked call, the user receives a busy tone, and the IP phone displays the message, “Park Slot Unavailable”.

Ensure that the user dials the retrieval prefix followed by the directed call park number.

User Cannot Park Calls

User cannot park calls. After the Transfer softkey (or Transfer button if available) is pressed and the directed call park number is dialed, the call does not get parked.

Ensure that the partition that is assigned to the call park number matches the partition that is assigned to the phone directory number. Ensure that the partition and calling search space are configured correctly for the device. For more information about the partition, see the *System Configuration Guide for Cisco Unified Communications Manager*.

User Receives a Reorder Tone After the Reversion Timer Expires

User cannot park calls. The user receives a reorder tone after the reversion timer expires.

Ensure that the user presses the Transfer softkey (or Transfer button if available) before dialing the directed call park number, and then presses the Transfer softkey (or Transfer button) again or goes on hook after dialing the directed call park number. Because directed call park is a transfer function, the directed call park number cannot be dialed alone.



Note You can complete the transfer only by going on hook rather than pressing the Transfer softkey (or Transfer button) a second time if the Transfer On-hook Enabled service parameter is set to True.

User Receives a Reorder Tone or Announcement

User cannot park calls. After pressing the Transfer softkey (or Transfer button if available) and dialing the directed call park number, the user receives a reorder tone or announcement.

Ensure that the dialed number is configured as a directed call park number.

User Cannot Park a Call at a Number Within The Range

After configuring a range of directed call park numbers, the user cannot park a call at a number within the range.

Review the syntax for entering a range of directed call park numbers. If incorrect syntax is used, the system may appear to configure the range when it actually does not.

Parked Calls Revert Too Quickly

Parked calls revert too quickly.

Set the Call Park Reversion Timer to a longer duration.

Park Slot Unavailable

User cannot park calls. After pressing the Transfer softkey (or Transfer button if available) and dialing the directed call park number, the user receives a busy tone, and the IP phone displays the message, “Park Slot Unavailable".

Ensure that the dialed directed call park number is not already occupied by a parked call or park the call on a different directed call park number.

Parked Calls Do Not Revert to the Parked Call Number

Parked calls do not revert to the number that parked the call.

Check the configuration of the directed call park number to ensure that it is configured to revert to the number that parked the call rather than to a different directory number.

Number or Range Cannot Be Deleted Because It Is in Use

When an attempt is made to delete a directed call park number or range, a message displays that indicates that the number or range cannot be deleted because it is in use.

You cannot delete a directed call park number that a device is configured to monitor (by using the BLF button). To determine which devices are using the number, click the Dependency Records link in the Directed Call Park Configuration window.



CHAPTER 32

Extension Mobility

- Extension Mobility Overview, on page 403
- Extension Mobility Prerequisites, on page 404
- Extension Mobility Configuration Task Flow, on page 404
- Cisco Extension Mobility Interactions, on page 413
- Cisco Extension Mobility Restrictions, on page 415
- Extension Mobility Troubleshooting, on page 416

Extension Mobility Overview

Cisco Extension Mobility allows users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones within your system. If you have a single phone that will be used by multiple workers, for example, you can configure extension mobility so that individual users can log in to the phone and access their settings without affecting settings on other user accounts.

After a user logs in using extension mobility and if the extension mobility profile is already associated to the application user, then CTI application sends device-related information. CTI application can control a device the user is logged into (using that extension mobility profile) without having to have direct control of the device. Therefore, the recording with the device profile association to the application user should work though they have not associated the device directly.

On authentication, if the login profile matches the login device (that is, the user has a user device profile that is configured for a Cisco IP Phone 7960 and logs in to a Cisco IP Phone 7960), Extension Mobility behaves the same way as it does with the older Unified CM versions:

- The phone automatically reconfigures with the individual user device profile information.

  If the user has one user device profile, then the system uses this profile. If the user has more than one user device profile, the user can choose the user device profile that will be used from a list.

- The user can access all the services that the user configured on the device profile.

  If that same user logs into a Cisco IP Phone model where the user does not have a configured user device profile, the login profile will not match the login device on authentication. In this scenario, the system loads the device profile default for that phone model onto the phone, and Extension Mobility works as described here:

  - The system copies all device-independent configuration (that is, user hold audio source, user locale, userid, speeddials, and directory number configuration except for the setting "line setting for this device") from the user device profile to the login device.

- The system uses the device profile default for that phone model for phone template and softkey template configuration and, if the phone can support addon modules, for the addon module.
- If the phone model supports Cisco IP Phone Services and they are configured, the system copies the services from the user device profile.

  If the user device profile does not have Cisco IP Phone Services configured, the system uses the Cisco IP Phone Services that are configured in the device profile default for the login device that is accessed during login. If parameters exist for the subscriber service, the system copies the parameters from the device profile default and the parameters may not reflect the correct information.

  For example, the following scenarios occur when a user who has a user device profile that is configured for Cisco IP Phone Model 7960 logs in to a Cisco IP Phone Model 7905, and the device default profile is loaded on the phone.

- The user can access the user's hold audio source, user locale, userid, speed-dials, and directory number configuration. The user cannot access his phone line setting; the system configured the phone line setting from the device profile default that is configured for the Cisco IP Phone 7905.
- The user can access the phone template and the softkey template of the Cisco IP Phone 7905.
- The user cannot access an addon module because Cisco IP Phone 7905 does not support it.
- The user can access Cisco IP Phone Services if they are configured for the Cisco IP Phone 7905, but the parameters from the subscriber services reflect the device profile default, not the parameters that the user chose on the User Options window.

  Users log out of Cisco Extension Mobility by pressing the Services button and choosing logout. If users do not log out themselves, the system will automatically log them out if you configured the Service Parameters to do so, or the next user of the phone can log out the previous user. After logout, Unified CM sends the logout profile to the phone and restarts the phone.

Extension Mobility Prerequisites

- A TFTP server that is reachable.
- Extension mobility functionality extends to most Cisco Unified IP Phones. Check the phone documentation to verify that Cisco Extension Mobility is supported.

Extension Mobility Configuration Task Flow

Before you begin

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the extension mobility feature.

	Command or Action	Purpose
Step 2	Activate Extension Mobility Services, on page 405	
Step 3	Configure the Cisco Extension Mobility Phone Service, on page 406	Configure the extension mobility IP phone service to which users can later subscribe to access extension mobility.
Step 4	Create an Extension Mobility Device Profile for Users, on page 407	Configure an extension mobility device profile. This profile acts as a virtual device that maps onto a physical device when a user logs in to extension mobility. The physical device takes on the characteristics in this profile.
Step 5	Associate a Device Profile to a User, on page 407	Associate a device profile to users so that they can access their settings from a different phone. You associate a user device profile to a user in the same way that you associate a physical device.
Step 6	Subscribe to Extension Mobility, on page 408	Subscribe IP phones and device profiles to the extension mobility service so that users can log in, use, and log out of extension mobility.
Step 7	Configure the Change Credential IP Phone Service, on page 408	To allow users to change their PINs on their phones, you must configure the change credential Cisco Unified IP Phone service and associate the user, the device profile, or the IP phone with the change credential phone service.
Step 8	(Optional) Configure Service Parameters for Extension Mobility, on page 409	If you want to modify the behavior of extension mobility, configure the service parameters.

Activate Extension Mobility Services

Procedure

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

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

Step 3 Activate the following services:

a) Cisco CallManager
b) Cisco Tftp
c) Cisco Extension Mobility
d) ILS Service

Note
You must choose publisher node to activate the ILS services.

Step 4 Click **Save**.

Step 5 Click **OK**.

Configure the Cisco Extension Mobility Phone Service

Configure the extension mobility IP phone service to which users can later subscribe to access extension mobility.

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.

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 Unified Communications Manager where Cisco Extension Mobility is activated and running.

It should be a IPv4 address.

Example:

`http://123.45.67.89:8080/emapp/EMAppServlet?device=#DEVICENAME#`

Example:

`http://[2001:0001:0001:0067:0000:0000:0000:0134]:8080/emapp/EMAppServlet?device=#DEVICENAME#`

This format allows a user to sign-in using User ID and PIN. You can configure more sign-in options for IP phone users who have subscribed to the extension mobility service. To configure more sign-in options, append the `loginType` parameter to the Service URL, in the following formats:

- `loginType=DN` enables users to sign in using Primary Extension and PIN.

  The Service URL format is:`http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=DN`.

- `loginType=SP` enables users to sign in using Self Service User ID and PIN.

  The Service URL format is:`http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=SP`.

- `loginType=UID` enables users to sign in using User ID and PIN.

  The Service URL format is:`http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=UID`.

If you do not append `loginType` to the end of the URL, the default sign-in option displayed is User ID and PIN.

Step 5 In the **Service Type** field, choose whether the service is provisioned to the Services, Directories, or Messages button.

Step 6 Click **Save**.

Create an Extension Mobility Device Profile for Users

Configure an extension mobility device profile. This profile acts as a virtual device that maps onto a physical device when a user logs in to extension mobility. The physical device takes on the characteristics in this profile.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings and choose an existing device profile from the resulting list.
- Click **Add New** to add a new device profile and choose an option from the **Device Profile Type**. Click **Next**.
- Choose a device protocol from the **Device Protocol** drop-down list and click **Next**.

Step 3 Configure the fields. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Step 5 From the **Association Information** section, click **Add a new DN**.

Step 6 In the **Directory Number** field, enter the directory number and click **Save**.

Step 7 Click **Reset** and follow the prompts.

Associate a Device Profile to a User

Associate a device profile to users so that they can access their settings from a different phone. You associate a user device profile to a user in the same way that you associate a physical device.

Tip You can use the Bulk Administration Tool (BAT) to add and delete several user device profiles for Cisco Extension Mobility at one time. See the Bulk Administration Guide for Cisco Unified Communications Manager.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing user, enter search criteria, and choosing an existing user from the resulting list.

- Click **Add New** to add a new user.

Step 3 Under **Extension Mobility**, locate the device profile that you created and move it from **Available Profiles** to **Controlled Profiles**.

Step 4 Check the **Home Cluster** check box.

Step 5 Click **Save**.

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.

Procedure

Step 1 Perform one of the following tasks from Cisco Unified CM Administration:

- Choose **Device** > **Phone**, specify search criteria, click **Find**, and choose a phone which users will use for extension mobility.
- Choose **Device** > **Device Settings** > **Device Profile**, specify search criteria, click **Find**, and choose the device profile that you created.

Step 2 From the **Related Links** drop-down list, choose **Subscribe/Unsubscribe Services**, and then 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**.

Step 6 Click **Save** and close the popup window.

Configure the Change Credential IP Phone Service

To allow users to change their PINs on their phones, you must configure the change credential Cisco Unified IP Phone service and associate the user, the device profile, or the IP phone with the change credential phone service.

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Service Name** field, enter **`Change Credential`**.

Step 4 In the **Service URL** field, enter the following value, where `server` designates the server where the Change Credential IP phone service runs:

`http://server:8080/changecredential/ChangeCredentialServlet?device=#DEVICENAME#`

Step 5 (Optional) In the **Secure-Service URL** field, enter the following value, where `server` is the server where the Change Credential IP phone service runs:

`https://server:8443/changecredential/ChangeCredentialServlet?device=#DEVICENAME#`

Step 6 Configure the remaining fields in the **IP Phone Services Configuration** window, and choose **Save**.

Step 7 To subscribe the Cisco Unified IP Phone to the Change Credential IP phone service, choose **Device** > **Phone**.

Step 8 In the **Phone Configuration** window, go to the **Related Links** drop-down list and choose **Subscribe/Unsubscribe Services**.

Step 9 Click **Go**.

Step 10 From the **Select a Service** drop-down list, choose the **Change Credential IP phone service**.

Step 11 Click **Next**.

Step 12 Click **Subscribe**.

Step 13 Click **Save**.

Configure Service Parameters for Extension Mobility

(Optional)

If you want to modify the behavior of extension mobility, configure the service parameters.

Procedure

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

Step 2 From the **Server** field, choose the node that is running the Cisco Extension Mobility service.

Step 3 From the **Service** field, choose **Cisco Extension Mobility**.

Step 4 Click **Advanced** to show all service parameters.

See Extension Mobility Service Parameters, on page 410 for more information about these service parameters and their configuration options.

Step 5 Click **Save**.

Extension Mobility Service Parameters

Table 32: Extension Mobility Service Parameters

Service Parameter	Description
Enforce Intra-cluster Maximum Login Time	Select **True** to specify a maximum time for local logins. After this time, the system automatically logs out the device. **False**, which is the default setting, means that no maximum time for logins exists. To set an automatic logout, you must choose **True** for this service parameter and also specify a system maximum login time for the **Intra-cluster Maximum Login Time** service parameter. Cisco Unified Communications Manager then uses the automatic logout service for all logins. If the value of **Enforce Intra-cluster Maximum Login Time** is set to **False** and you specify a valid maximum login time for the **Intra-cluster Maximum Login Time** service parameter, then the value of **Enforce Intra-cluster Maximum Login Time** automatically changes to **True**.
Intra-cluster Maximum Login Time	This parameter sets the maximum time that a user can be locally logged in to a device, such as 8:00 (8 hours) or :30 (30 minutes). The system ignores this parameter and set the maximum login time to 0:00, if the **Enforce Intra-cluster Maximum Login Time** parameter is set to **False**. Valid values are between 0:00 and 168:00 in the format HHH:MM, where HHH represents the number of hours and MM represents the number of minutes. **Note** If you grant a user access to set their Extension Mobility maximum login time (configured via the **Allow End User to set their Extension Mobility maximum login time** check box in the **User Profile Configuration**) the user's configuration in the Self-Care Portal overrides the value of the **Intra-cluster Maximum Login Time** service parameter.
Maximum Concurrent Requests	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.

Service Parameter	Description
Multiple Login Behavior	When users are logged in to one phone and then login to a second phone either in the same cluster or on a different cluster, users can view the login behavior on the second phone based on the **Multiple Login Behavior** setting defined on the **Service Parameter Configuration** page. Choose one of the following options from the drop–down list: • Multiple Logins Allowed—You can login to more than one device at a time. • Multiple Logins Not Allowed—You can be logged in to only one device. The login attempts to the second device fails and the phone displays the error code "25" (Multi-Login Not Allowed). You can login successfully, only when you have logged out from the first device. This is the default value. • Auto Logout—When you try to login to a second device (either Extension Mobility or Extension Mobility Cross Cluster), the Cisco Unified Communications Manager automatically logs you out of the first device. This is a required field. **Note** Multiple login behavior is also applicable between two Extension Mobility Cross Cluster logins.
Alphanumeric User ID	Choose **True** to allow the user ID to contain alphanumeric characters. Choosing **False** allows the user ID to contain only numeric characters. **Note** The Alphanumeric User ID parameter applies systemwide. You can have a mix of alphanumeric and numeric user IDs. The system supports only user IDs that can be entered by using the alphanumeric keypad. The case-sensitive userid field requires the characters to be lowercase.
Remember the Last User Logged In	When you choose **False**, the system does not remember the last user who logged in to the phone. Use this option when the user access the phone on a temporary basis only. Choose True to remember the last user that logged into the phone. Use this option when a phone has only one user. For example, Cisco Extension Mobility is used to enable the types of calls that are allowed from a phone. Individuals who are not logged in and who are using their office phone can make only internal or emergency calls. But after logging in using Cisco Extension Mobility, the user can make local, long-distance, and international calls. In this scenario, only this user regularly logs in to the phone. It makes sense to set the Cisco Extension Mobility to remember the last user ID that logged in.

Service Parameter	Description
Clear Call Logs on Intra-cluster EM	Choose **True** to specify that the call logs are cleared during the Cisco Extension Mobility manual login and logout process. While a user is using the Cisco Extension Mobility service on an IP phone, all calls (placed, received, or missed) appear in a call log and can be retrieved and seen on the IP phone display. To ensure privacy, set the Clear Call Log service parameter to **True**. This ensures that the call logs are cleared when a user logs out and another user logs in. For extension mobility cross cluster (EMCC), the call log is always cleared when the user logs in or out of a phone. **Note** Call logs are cleared only during manual login/logout. If a Cisco Extension Mobility logout occurs automatically or any occurrence other than a manual logout, the call logs are not cleared.
Validate IP Address	This parameter sets whether validation occurs on the IP address of the source that is requesting login or logout. If the parameter is set to **True**, the IP address from which a Cisco Extension Mobility log in or log out request occurs and is validated to ensure that it is trusted. Validation is first performed against the cache for the device that will log in or log out. If the IP address is found in the cache or in the list of trusted IP addresses or is a registered device, the device can log in or log out. If the IP address is not found, the log in or log out attempt is blocked. If the parameter is set to **False**, the Cisco Extension Mobility log in or log out request is not validated. Validation of IP addresses can affect the time that is required to log in or log out a device, but it offers additional security that prevents unauthorized log in or log out attempts. This function is recommended, especially when used with logins from separate trusted proxy servers for remote devices.
Trusted List of IPs	This parameter appears as a text box (the maximum length is 1024 characters). You can enter strings of trusted IP addresses or hostnames which are separated by semicolons, in the text box. IP address ranges and regular expressions are not supported.
Allow Proxy	If the parameter is **True**, the Cisco Extension Mobility log in and log out operations that use a web proxy are allowed. If the parameter is **False**, the Cisco Extension Mobility log in and log out requests coming from behind a proxy get rejected. The setting that you select takes effect only if the Validate IP Address parameter specifies true.

Service Parameter	Description
Extension Mobility Cache Size	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. The value that you enter takes effect only if the Validate IP Address parameter is **True**.

Cisco Extension Mobility Interactions

Table 33: Cisco Extension Mobility Interactions

Feature	Interaction
Assistant	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).
BLF Presence	When you configure BLF/speed dial buttons in a user device profile, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF/SpeedDial buttons after you log in to the device. When the extension mobility user logs out, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF/SpeedDial buttons for the logout profile that is configured.
Call Display Restrictions	When you enable call display restrictions, Cisco Extension Mobility functions as usual: when a user is logged in to the device, the presentation or restriction of the call information depends on the user device profile that is associated with that user. When the user logs out, the presentation or restriction of the call information depends on the configuration that is defined for that phone type in the Phone Configuration window. To use call display restrictions with Cisco Extension Mobility, check the **Ignore Presentation Indicators (internal calls only)** check obx 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

Feature	Interaction
Do Not Disturb	For extension mobility, the device profile settings include do not disturb (DND) incoming call alert and DND status. When a user logs in and enables DND, the DND incoming call alert and DND status settings are saved, and these settings are used when the user logs in again. **Note** When a user who is logged in to extension mobility modifies the DND incoming call alert or DND status settings, this action does not affect the actual device settings.
Intercom	Cisco Extension Mobility supports the intercom feature. To support intercom, Cisco Extension Mobility uses a default device that is configured for an intercom line. An intercom line is presented on only the default device. You can assign an intercom line to a device profile. When a user logs in to a device that is not the default device, the 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 IPv6. You can use phones with an IP addressing mode of IPv6 or dual-stack (IPv4 and IPv6).
Prime Line	If you select **On** for the **Always Use Prime Line** parameter in the **Device Profile** or **Default Device Profile Configuration** window, a Cisco Extension Mobility user can use this feature after logging in to the device that supports Cisco Extension Mobility.

Cisco Extension Mobility Restrictions

Note When database replication within the Cisco Unified Communications Manager cluster is not functioning per expectation, it impacts the extension mobility functionality. To improve resiliency, we recommend the following:

- Configure the backup CCM service, TFTP service, and extension mobility service in an alternate data center.
- Use a load balancer to send the extension mobility login request to the extension mobility service in different data centers in a round robin method such that the extension mobility service in another data center can handle the requests when the first is unreachable.
- Use the backup extension mobility URL to log in when the primary URL fails.

Table 34: Cisco Extension Mobility Restrictions

Feature	Restriction
Cache	Cisco Extension Mobility maintains a cache of all logged-in user information for 2 minutes. If a request comes to extension mobility about 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.
Call Back	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 canceled.
Character Display	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.
Hold Reversion	Cisco Extension Mobility does not support the hold reversion feature.
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 CM Administration.
Secure Tone	Cisco Extension Mobility and join across line services are disabled on protected phones.

Feature	Restriction
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.
Remember the Last User Logged In	The service parameter **Remember the Last User Logged In** is applicable only for default Extension Mobility service URL or the Extension Mobility service URL with `loginType` as `UID`.

Extension Mobility Troubleshooting

Troubleshoot Extension Mobility

Procedure

- Configure the Cisco Extension Mobility trace directory and enable debug tracing by performing the following steps:
  a) From Cisco Unified Serviceability, choose **Trace** > **Trace Configuration**.
  b) From the **Servers** drop-down list, select a server.
  c) From the **Configured Services** drop-down-list, select **Cisco Extension Mobility**.
- Make sure that you entered the correct URL for the Cisco Extension Mobility service. Remember that the URL is case sensitive.
- Check that you have thoroughly and correctly performed all the configuration procedures.
- If a problem occurs with authentication of a Cisco Extension Mobility user, go to the user pages and verify the PIN.

Authentication Error

Problem "Error 201 Authentication Error" appears on the phone.

Solution The user should check that the correct user ID and PIN were entered; the user should check with the system administrator that the user ID and PIN are correct.

Blank User ID or PIN

Problem "Error 202 Blank User ID or PIN" appears on the phone.

Solution Enter a valid user ID and PIN.

Busy Please Try Again

Problem "Error 26 Busy Please Try Again" appears on the phone.

Solution Check whether the number of concurrent login and logout requests is greater than the **Maximum Concurrent requests** service parameter. If so, lower the number of concurrent requests.

Note To verify the number of concurrent login and logout requests, use the Cisco Unified Real-Time Monitoring Tool to view the Requests In Progress counter in the Extension Mobility object. For more information, see the Cisco Unified Real-Time Monitoring Tool Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html

Database Error

Problem "Error 6 Database Error" appears on the phone.

Solution Check whether a large number of requests exists. If a large number of requests exists, the Requests In Progress counter in the Extension Mobility object counter shows a high value. If the requests are rejected because of a large number of concurrent requests, the Requests Throttled counter also shows a high value. Collect detailed database logs.

Dev Logon Disabled

Problem "Error 22 Dev Logon Disabled" appears on the phone.

Solution Verify that you checked the **Enable Extension Mobility** check box in the **Phone Configuration** window (**Device** > **Phone**).

Device Name Empty

Problem "Error 207 Device Name Empty" appears on the phone.

Solution Check that the URL that is configured for Cisco Extension Mobility is correct. See the Related Topics section for more information.

Related Topics

Configure the Cisco Extension Mobility Phone Service, on page 406

EM Service Connection Error

Problem "Error 207 EM Service Connection Error" appears on the phone.

Solution Verify that the Cisco Extension Mobility service is running by selecting **Tools** > **Control Center—Feature** in Cisco Unified Serviceability.

Extension Mobility Performance During Upgrade

Problem Extension Mobility (EM) login performance during Publisher switch version after the upgrade.

Solution If Extension Mobility (EM) users are logged in during the switch version upgrade of Unified Communications Manager Publisher, and if the Publisher is inactive, EM login data is lost during the switch version and EM profiles are logged out.

Note If EM login profiles are logged out, users can log in again, or log in only when Unified Communications Manager is active after the switch version.

Host Not Found

Problem The "Host Not Found" error message appears on the phone.

Solution Check that the Cisco Tomcat service is running by selecting **Tools** > **Control Center—Network Services** in CIsco Unified Serviceability.

HTTP Error

Problem HTTP Error (503) appears on the phone.

Solution

- If you get this error when you press the **Services** button, check that the Cisco IP Phone Services service is running by selecting **Tools** > **Control Center—Network Services** in Cisco Unified Serviceability.
- If you get this error when you select Extension Mobility service, check that the Cisco Extension Mobility Application service is running by selecting **Tools** > **Control Center—Network Services** in Cisco Unified Serviceability.

Phone Resets

Problem After users log in or log out, their phones reset instead of restarting.

Possible Cause Locale change is the probable cause of the reset.

Solution No action is required. If the user locale that is associated with the logged-in user or profile is not the same as the locale or device, after a successful login the phone will restart and then reset. This pattern occurs because the phone configuration file is rebuilt.

Phone Services Unavailable After Login

Problem After logging in, the user finds that the phone services are not available.

Possible Cause This problem occurs because the user profile had no services associated with it when it was loaded on the phone.

Solution

- Ensure that the user profile includes the Cisco Extension Mobility service.
- Change the configuration of the phone where the user is logged in to include Cisco Extension Mobility. After the phone is updated, the user can access the phone services.

Phone Services Unavailable After Logout

Problem After a user logs out and the phone reverts to the default device profile, the phone services are no longer available.

Solution Subscribe the phone to the Cisco Extension Mobility service.

User Logged in Elsewhere

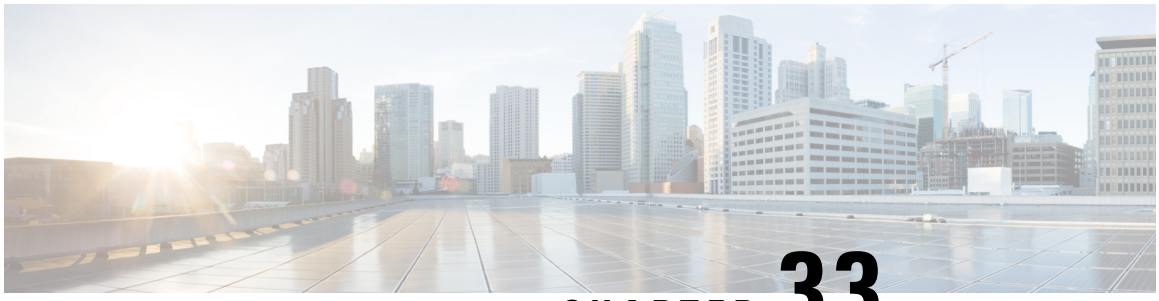
Problem "Error 25 User Logged in Elsewhere" appears on the phone.

Solution Check whether the user is logged in to another phone. If multiple logins must be allowed, ensure that the **Multiple Login Behavior** service parameter is set to **Multiple Logins Allowed**.

User Profile Absent

Problem "Error 205 User Profile Absent" appears on the phone.

Solution Associate a device profile to the user.



CHAPTER 33

Extension Mobility Cross Cluster

- Extension Mobility Cross Cluster Overview, on page 421
- Extension Mobility Cross Cluster Prerequisites, on page 421
- Extension Mobility Cross Cluster Configuration Task Flow, on page 422
- Extension Mobility Cross Cluster Interactions, on page 443
- Extension Mobility Cross Cluster Restrictions, on page 444
- Extension Mobility Cross Cluster Troubleshooting, on page 448

Extension Mobility Cross Cluster Overview

The extension mobility cross cluster (EMCC) feature provides users with the same functionality as extension mobility, but also allows them to move from one cluster (the home cluster) and log in to a temporary phone on another remote cluster (the visiting cluster). From there, they can access their phone settings from any location as if they were using an IP Phone at the home office.

Note Ensure that the remote EMCC cluster support the minimum TLS version configured on your home cluster.

Extension Mobility Cross Cluster Prerequisites

- Other call-control entities that support and use the extension mobility cross cluster (EMCC) configuration; for example, other Cisco Unified Communications Manager clusters, EMCC intercluster service profiles, and EMCC remote cluster services
- Clusters that are set to nonsecure or mixed mode. See Extension Mobility Cross Cluster and Security Mode for Different Cluster Versions, on page 446 for more information.
- Supported phones in secure or nonsecure mode

Extension Mobility Cross Cluster Configuration Task Flow

Before you begin

- Review Extension Mobility Cross Cluster Prerequisites, on page 421
- Review **Extension Mobility Cross Cluster Interaction and Restriction**

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the extension mobility cross cluster feature.
Step 2	To Configure Extension Mobility, on page 423, perform the following subtasks: • Activate Services for Extension Mobility Cross Cluster, on page 424 • Configure the Extension Mobility Phone Service, on page 424 • Configure a Device Profile for Extension Mobility Cross Cluster, on page 425 • Enable Extension Mobility Cross Cluster for a User, on page 431 • Subscribe Devices to Extension Mobility, on page 432	Configure extension mobility to allow users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones in one cluster. Perform this task flow on both home and remote clusters, so that users will be able to access settings from either a home or visiting cluster.
Step 3	To Configure Certificates for Extension Mobility Cross Cluster, on page 432, perform the following subtasks: • Activate the Bulk Provisioning Service, on page 433 • Configure Bulk Certificate Management and Export Certificates, on page 433 • Consolidate the Certificates, on page 434 • Import the Certificates into the Clusters, on page 434	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.
Step 4	To Configure Extension Mobility Cross Cluster Devices and Templates, on page 436, perform the following subtasks: • Create a Common Device Configuration, on page 436 • Configure an Extension Mobility Cross Cluster Template, on page 436	

	Command or Action	Purpose
	• Set the Default Template, on page 437 • Add Extension Mobility Cross Cluster Devices, on page 437	
Step 5	Configure a Geolocation Filter for Extension Mobility Cross Cluster, on page 438	Configure a geolocation filter to specify criteria for device location matching, such as country, state, and city values. Geolocations are used to identify the location of a device, and the filter indicates what parts of the geolocation are significant.
Step 6	Configure Feature Parameters for Extension Mobility Cross Cluster, on page 438	Select values for the feature parameters that you configured, such as the geolocation filter.
Step 7	Configure Intercluster SIP Trunk for Extension Mobility Cross Cluster, on page 442	Configure trunks to process inbound or outbound traffic for intercluster PSTN access and RSVP agent services. You can configure one trunk for both PSTN access and RSVP agent services or one trunk for each service. You do not need more than two SIP trunks for extension mobility cross cluster.
Step 8	Configure an Intercluster Service Profile for Extension Mobility Cross Cluster, on page 442	Configure the intercluster service profile to activate extension mobility cross cluster. The profile collects all the configuration that precedes and provides a results report.
Step 9	Configure Remote Cluster Services, on page 443	Configure the remote cluster for extension mobility cross cluster. This step completes the link between the home cluster with remote (visiting) cluster.

Configure Extension Mobility

Configure extension mobility to allow users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones in one cluster. Perform this task flow on both home and remote clusters, so that users will be able to access settings from either a home or visiting cluster.

Procedure

	Command or Action	Purpose
Step 1	Activate Services for Extension Mobility Cross Cluster, on page 424	
Step 2	Configure the Extension Mobility Phone Service, on page 424	Create the Extension Mobility phone service to which you can subscribe your users.

	Command or Action	Purpose
Step 3	Configure a Device Profile for Extension Mobility Cross Cluster, on page 425	Create a device profile to map settings onto a real device when a user logs in to Extension Mobility cross cluster.
Step 4	Enable Extension Mobility Cross Cluster for a User, on page 431	
Step 5	Subscribe Devices to Extension Mobility, on page 432	Enable Extension Mobility on devices and subscribe to the service if you have not set up an enterprise subscription for all devices.

Activate Services for Extension Mobility Cross Cluster

Procedure

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

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

Step 3 Activate the following services:

a) Cisco CallManager
b) Cisco Tftp
c) Cisco Extension Mobility
d) ILS Service

Note
You must choose publisher node to activate the ILS services.

Step 4 Click **Save**.

Step 5 Click **OK**.

Configure the Extension Mobility Phone Service

Create the Extension Mobility phone service to which you can subscribe your users.

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 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 secure service URL in the following format:

`https://<IP Address>:8443/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#`

Step 6 (Optional) If you want to configure more sign-in options, append the `loginType` parameter to the Service URL in the following formats:

- `loginType=DN` enables users to sign in using Primary Extension and PIN.The Service URL format is: `http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#&loginType=DN`.
- `loginType=SP` enables users to sign in using Self Service User ID and PIN.

  The Service URL format is: `http: //<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#&loginType=SP`.
- `loginType=UID` enables users to sign in using User ID and PIN.

  The Service URL format is: `http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&EMCC=#EMCC#&loginType=UID`.

The `loginType` parameter can also be appended to a secure URL. If you do not append `loginType` to the end of the URL, the default sign in option displayed is User ID and PIN.

Step 7 Use the default values for the **Service Category** and **Service Type** fields.

Step 8 Check the **Enable** check box.

Step 9 (Optional) Check the **Enterprise Subscription** check box to subscribe all phones and device profiles to this phone service.

Note
If you check this check box when configuring the service for the first time, you will set up this IP phone service as an enterprise subscription service. All phones and device profiles in the enterprise will automatically subscribe to this IP phone service, removing the need for you to subscribe them individually.

Step 10 Click **Save**.

Configure a Device Profile for Extension Mobility Cross Cluster

Create a device profile to map settings onto a real device when a user logs in to Extension Mobility cross cluster.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify an existing device profile, enter search criteria. Click a device profile name in the resulting list.

- Click **Add New** to add a new device profile and click **Next** to choose a device profile type. Click **Next** to choose a protocol, then click **Next**.

Step 3 Configure the fields on the **Device Profile Configuration** window. See Device Profile Fields for Extension Mobility Cross Cluster, on page 426 for more information about the fields and their configuration options.

Step 4 Click **Save**.

Step 5 Add a directory number (DN) to the new device profile.

Device Profile Fields for Extension Mobility Cross Cluster

Table 35: Device Profile Settings

Field	Description
Product Type	Displays the product type to which this device profile applies.
Device Protocol	Displays the device protocol to which this device profile applies.
Device Profile Name	Enter a unique name. This name can comprise up to 50 characters in length.
Description	Enter a description of the device profile. For text, use anything that describes this particular user device profile.
User Hold MOH Audio Source	Specifies the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list. If you do not choose an audio source, Unified Communications Manager uses the audio source that is defined in the device pool or the system default if the device pool does not specify an audio source ID. **Note** You define audio sources in the Music On Hold Audio Source Configuration window. For access, choose **Media Resources** > **Music On Hold Audio Source**.
User Locale	From the drop-down list, choose the locale that is associated with the phone user interface. The user locale identifies a set of detailed information, including language and font, to support users. Unified Communications Manager makes this field available only for phone models that support localization. **Note** If no user locale is specified, Unified Communications Manager uses the user locale that is associated with the device pool. If the users require information to display (on the phone) in any language other than English, verify that the locale installer is installed before configuring user locale. See the Unified Communications Manager Locale Installer documentation.

Field	Description
Phone Button Template	From the Phone Button Template drop-down list, choose a phone button template. **Tip** If you want to configure BLF/SpeedDials for the profile for presence monitoring, choose a phone button template that you configured for BLF/SpeedDials. After you save the configuration, the Add a New BLF SD link displays in the Association Information pane. For more information on BLF/SpeedDials, see the Feature Configuration Guide for Cisco Unified Communications Manager.
Softkey Template	From the Softkey Template drop-down list, choose the softkey template from the list that displays.
Privacy	From the Privacy drop-down list, choose On for each phone on which you want privacy. For more information, see the Feature Configuration Guide for Cisco Unified Communications Manager.
Single Button Barge	From the drop-down list, choose from the following options: • Off—This device does not allow users to use the Single Button Barge/cBarge feature. • Barge—Choosing this option allows users to press the Single Button Barge shared-line button on the phone to barge into a call using Barge. • Default—This device inherits the Single Button Barge/cBarge setting from the service parameter and device pool settings. **Note** If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting. For more information, see the Feature Configuration Guide for Cisco Unified Communications Manager.
Join Across Lines	From the drop-down list, choose from the following options: • Off—This device does not allow users to use the Join Across Lines feature. • On—This device allows users to join calls across multiple lines. • Default—This device inherits the Join Across Lines setting from the service parameter and device pool settings. **Note** If the server parameter and device pool settings are different, the device will inherit the setting from the service parameter setting. For more information, see the System Configuration Guide for Cisco Unified Communications Manager.

Field	Description
Always Use Prime Line	From the drop-down list, choose one of the following options: • Off—When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received. • On—When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls. • Default—Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco CallManager service.
Always Use Prime Line for Voice Message	From the drop-down list, choose one of the following options: • On—If the phone is idle, the primary line on the phone becomes the active line for retrieving voice messages when the phone user presses the Messages button on the phone. • Off—If the phone is idle, pressing the Messages button on the phone automatically dials the voice-messaging system from the line that has a voice message. Unified Communications Manager always selects the first line that has a voice message. If no line has a voice message, the primary line gets used when the phone user presses the Messages button. • Default—Unified Communications Manager uses the configuration from the Always Use Prime Line for Voice Message service parameter, which supports the Cisco CallManager service.
Ignore Presentation Indicators (internal calls only)	To configure call display restrictions and ignore any presentation restriction that is received for internal calls, check the “Ignore Presentation Indicators (internal calls only)” check box. **Tip** Use this configuration in combination with the calling line ID presentation and connected line ID presentation configuration at the translation pattern level. Together, these settings allow you to configure call display restrictions to selectively present or block calling and/or connected line display information for each call. For more information about call display restrictions, see the Feature Configuration Guide for Cisco Unified Communications Manager.
Do Not Disturb	Check this check box to enable Do Not Disturb.

Field	Description
DND Option	When you enable DND on the phone, this parameter allows you to specify how the DND feature handles incoming calls: • Call Reject—This option specifies that no incoming call information gets presented to the user. Depending on how you configure the DND Incoming Call Alert parameter, the phone may play a beep or display a flash notification of the call. • Ringer Off—This option turns off the ringer, but incoming call information gets presented to the device, so that the user can accept the call. • Use Common Phone Profile Setting—This option specifies that the DND Option setting from the Common Phone Profile window will get used for this device. **Note** For 7940/7960 phones that are running SCCP, you can only choose the Ringer Off option. For mobile devices and dual-mode phones, you can only choose the Call Reject option. When you activate DND Call Reject on a mobile device or dual-mode phone, no call information gets presented to the device.
DND Incoming Call Alert	When you enable the DND Ringer Off or Call Reject option, this parameter specifies how a call displays on a phone. From the drop-down list, choose one of the following options: • None—This option specifies that the DND Incoming Call Alert setting from the Common Phone Profile window will get used for this device. • Disable—This option disables both beep and flash notification of a call but for the DND Ringer Off option, incoming call information still gets displayed. For the DND Call Reject option, no call alerts display and no information gets sent to the device. • Beep Only—For an incoming call, this option causes the phone to play a beep tone only. • Flash Only—For an incoming call, this option causes the phone to display a flash alert.
Extension Mobility Cross Cluster CSS	From the drop-down list, choose an existing Calling Search Space (CSS) to use for this device profile for the Extension Mobility Cross Cluster feature. (To configure a new CSS or modify an existing CSS, choose **Call Routing** > **Class of Control** > **Calling Search Space** in Unified Communications Manager.) Default value specifies None. The home administrator specifies this CSS, which gets used as the device CSS that gets assigned to the phone when the user logs in to this remote phone. For more information, see the Feature Configuration Guide for Cisco Unified Communications Manager.

Field	Description
Module 1	You can configure one or two expansion modules for this device profile by choosing phone templates from the expansion module drop-down list in the expansion module fields. **Note** You can view a phone button list at any time by choosing the View button list link next to the phone button template fields. A separate dialog box pops up and displays the phone buttons for that particular expansion module. Choose the appropriate expansion module or None.
Module 2	Choose the appropriate expansion module or None.
MLPP Domain	If this user device profile will be used for MLPP precedence calls, choose the MLLP Domain from the drop-down list. **Note** You define MLPP domains in the MLPP Domain Configuration window. For access, choose **System** > **MLPP Domain**.
MLPP Indication	If this user device profile will be used for MLPP precedence calls, assign an MLPP Indication setting to the device profile. This setting specifies whether a device that can play precedence tones will use the capability when it places an MLPP precedence call. From the drop-down list, choose a setting to assign to this device profile from the following options: 1. Default—This device profile inherits its MLPP indication setting from the device pool of the associated device. 2. Off—This device does not handle nor process indication of an MLPP precedence call. 3. On—This device profile does handle and process indication of an MLPP precedence call. **Note** Do not configure a device profile with the following combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.

Field	Description
MLPP Preemption	If this user device profile will be used for MLPP precedence calls, assign an MLPP Preemption setting to the device profile. This setting specifies whether a device that can preempt calls in progress will use the capability when it places an MLPP precedence call. From the drop-down list, choose a setting to assign to this device profile from the following options: 1. Default—This device profile inherits its MLPP preemption setting from the device pool of the associated device. 2. Disabled—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls. 3. Forceful—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls. **Note** Do not configure a device profile with the following combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.
Login User Id	From the Login User ID drop-down list, choose a valid login user ID. **Note** If the device profile is used as a logout profile, specify the login user ID that will be associated with the phone. After the user logs out from this user device profile, the phone will automatically log in to this login user ID.

Enable Extension Mobility Cross Cluster for a User

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing user and choosing an existing user from the resulting list.
- Click **Add New** to add a new user.

Step 3 In the **Extension Mobility** pane, check the **Enable Extension Mobility Cross Cluster** check box.

Step 4 Choose the device profile from the **Available Profiles** list pane in the **Extension Mobility** pane.

Step 5 Move the device profile to the **Controlled Profiles** list pane.

Step 6 Click **Save**.

Subscribe Devices to Extension Mobility

Enable Extension Mobility on devices and subscribe to the service if you have not set up an enterprise subscription for all devices.

Procedure

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

Step 2 Find the phone on which users can use Extension Mobility Cross Cluster.

Step 3 For this device, check the **Enable Extension Mobility** check box in the **Extension Information** pane.

Step 4 In the **Phone Configuration** window, choose the **Subscribe/Unsubscribe Services** option in the **Related Links** drop-down list.

Step 5 Click **Go**.

Step 6 In the popup window that opens, choose the **Extension Mobility** service in the **Select a Service** drop-down list.

Step 7 Click **Next**.

Step 8 Click **Subscribe**.

Step 9 From the popup window, click **Save**, and then close the window.

Step 10 In the **Phone Configuration** window, click **Save**.

Step 11 Click **OK** if prompted.

Configure Certificates for Extension Mobility Cross Cluster

To configure the home and remote clusters properly, you must export certificates on each cluster to the same SFTP server and SFTP directory and consolidate them on one of the participating clusters. This procedure ensures that trust is established between the two clusters.

Before you begin

Configure Extension Mobility, on page 423

Procedure

	Command or Action	Purpose
Step 1	Activate the Bulk Provisioning Service, on page 433	
Step 2	Configure Bulk Certificate Management and Export Certificates, on page 433	Configure bulk certificate management in Cisco Unified OS Administration to export the certificates from both the home and remote clusters.
Step 3	Consolidate the Certificates, on page 434	Consolidate certificates when all participating clusters have exported their certificates. This

	Command or Action	Purpose
		option is available only if two or more clusters exported their certificates to the SFTP server.
Step 4	Import the Certificates into the Clusters, on page 434	Import the certificates back into the home and remote (visiting) clusters.

Activate the Bulk Provisioning Service

Before you begin

Configure Extension Mobility, on page 423

Procedure

Step 1 From Cisco Unified Serviceability, choose **Tools** > **Service Activation**.
Step 2 From the **Server** drop-down list, choose the publisher node.
Step 3 Check the **Cisco Bulk Provisioning Service** check box.
Step 4 Click **Save**.
Step 5 Click **OK**.

Configure Bulk Certificate Management and Export Certificates

Configure bulk certificate management in Cisco Unified OS Administration to export the certificates from both the home and remote clusters.

This procedure exports the certificates to an SFTP server.

Note
- Every participating cluster must export certificates to the same SFTP server and SFTP directory.
- You must export certificates on the cluster whenever the Tomcat, Tomcat-ECDSA, TFTP, or CAPF certificates are regenerated on any of the cluster nodes.

Procedure

Step 1 From Cisco Unified OS Administration, choose **Security** > **Bulk Certificate Management**.
Step 2 Configure the settings for a TFTP server that both the home and remote clusters can reach. See the online help for information about the fields and their configuration options.
Step 3 Click **Save**.
Step 4 Click **Export**.
Step 5 In the **Bulk Certificate Export** window, choose **All** for the **Certificate Type** field.

Step 6 Click **Export**.

Step 7 Click **Close**.

Note
The preceding steps are performed when certificates are self-signed and there is no common trust in another cluster. If there is a common trust or the same signer, then the export of ALL certificates is not needed.

Consolidate the Certificates

Consolidate certificates when all participating clusters have exported their certificates. This option is available only if two or more clusters exported their certificates to the SFTP server.

This procedure consolidates all PKCS12 files in the SFTP server to form a single file.



Note If you export new certificates after consolidation, you must perform this procedure again to include the newly exported certificates.

Procedure

Step 1 From From Cisco Unified OS Administration, choose **Security** > **Bulk Certificate Management** > **Consolidate** > **Bulk Certificate Consolidate**.

Step 2 In the **Certificate Type** field, choose **All**.

Step 3 Click **Consolidate**.

Note
When the bulk certificate consolidate is performed, the certificates are then uploaded to the remote cluster as follows:

- CAPF certificate gets uploaded as a CallManager-trust
- Tomcat certificate gets uploaded as a Tomcat-trust
- CallManager certificate gets uploaded as a CallManager-trust
- CallManager certificate gets uploaded as a Phone-SAST-trust
- ITLRecovery certificate gets uploaded as a PhoneSast-trust and CallManager-trust

Import the Certificates into the Clusters

Import the certificates back into the home and remote (visiting) clusters.



Note After an upgrade, these certificates are preserved. You do not need to reimport or reconsolidate certificates.



Note Ensure that you retain all the files that are created to ensure smooth consolidation between different Unified Communications Manager versions.



Caution After you import the certificates, the phones on the cluster will automatically restart.

Procedure

Step 1 From From Cisco Unified OS 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**.

Note
When the bulk certificate import is performed, the certificates are then uploaded to the remote cluster as follows:

- CAPF certificate gets uploaded as a CallManager-trust
- Tomcat certificate gets uploaded as a Tomcat-trust
- CallManager certificate gets uploaded as a CallManager-trust and a Phone-SAST-trust
- ITLRecovery certificate gets uploaded as a Phone-SAST-trust and CallManager-trust

Note
The following types of certificates determine phones that are restarted:

- CallManager—All phones only if TFTP service is activated on the node that the certificate belongs.
- TVS—Some phones based on CallManager group membership.
- CAPF—All phones only if CAPF is activated.

Configure Extension Mobility Cross Cluster Devices and Templates

Procedure

	Command or Action	Purpose
Step 1	Create a Common Device Configuration, on page 436	Configure a common device configuration to specify the services or features that will be associated with a particular user.
Step 2	Configure an Extension Mobility Cross Cluster Template, on page 436	Create an extension mobility cross cluster template to link the common device configuration with this feature.
Step 3	Set the Default Template, on page 437	Set the extension mobility cross cluster template that you created as the default template.
Step 4	Add Extension Mobility Cross Cluster Devices, on page 437	Insert extension mobility cross cluster device entries into your system database. Each device is identified with a unique name in the format EMCC1, EMCC2, and so on. The Bulk Administration Tool assigns device numbers by obtaining the last one used.

Create a Common Device Configuration

Configure a common device configuration to specify the services or features that will be associated with a particular user.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify an existing common device configuration and choose a common device configuration from the resulting list.
- Click **Add New** to add a new common device configuration.

Step 3 Configure the fields on the **Common Device Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure an Extension Mobility Cross Cluster Template

Create an extension mobility cross cluster template to link the common device configuration with this feature.

Procedure

Step 1 From 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. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Set the Default Template

Set the extension mobility cross cluster template that you created as the default template.

Procedure

Step 1 From 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 extension mobility cross cluster 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.

Add Extension Mobility Cross Cluster Devices

Insert extension mobility cross cluster device entries into your system database. Each device is identified with a unique name in the format EMCC1, EMCC2, and so on. The Bulk Administration Tool assigns device numbers by obtaining the last one used.

Procedure

Step 1 From From Cisco Unified CM Administration, choose **Bulk Administration** > **EMCC** > **Insert/Update EMCC**.

Step 2 Click **Insert EMCC Devices**.

Step 3 Enter the number of 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 the 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 for Extension Mobility Cross Cluster

Configure a geolocation filter to specify criteria for device location matching, such as country, state, and city values. Geolocations are used to identify the location of a device, and the filter indicates what parts of the geolocation are significant.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Geolocation Filter**.

Step 2 Click **Add New**.

Step 3 Configure the fields on the **Geolocation Filter Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure Feature Parameters for Extension Mobility Cross Cluster

Select values for the feature parameters that you configured, such as the geolocation filter.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Advanced Features** > **EMCC** > **EMCC Feature Configuration**.

Step 2 Configure the fields on the **EMCC Feature Configuration** window. See Feature Parameter Fields for Extension Mobility Cross Cluster, on page 438 for more information about the fields and their configuration options.

Step 3 Click **Save**.

Feature Parameter Fields for Extension Mobility Cross Cluster

Table 36: Feature Parameter Fields for Extension Mobility Cross Cluster

EMCC Parameter	Description
Default TFTP Server for EMCC Login Device	Choose the computer name or IP address of the default TFTP server that devices logging into extension mobility cross cluster (EMCC) from a remote cluster should use.

EMCC Parameter	Description
Backup TFTP Server for EMCC Login Device	Choose the computer name or IP address of the backup TFTP server that devices logging into EMCC from a remote cluster should use.
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 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.
Enable All Remote Cluster Services When Adding A New Remote Cluster	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 Unified Communications Manager). 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.
CSS for PSTN Access SIP Trunk	Choose the calling search space (CSS) that the PSTN Access SIP trunk for processing EMCC calls uses. The PSTN Access SIP trunk is the SIP trunk that you configured for PSTN access in the **Intercluster Service Profile** window. Calls over this trunk are intended for and are routed to only the local PSTN that is co-located with the EMCC logged-in phone that initiates the call. Valid values are the following: • Use Trunk CSS (PSTN calls use the local route group, which can prove useful for properly routing emergency service calls) • Use phone's original device CSS (PSTN calls are routed using the configured calling search space on the remote phone, that is, the CSS that is used when the phone is not logged into EMCC). The default value is Use trunk CSS.

EMCC Parameter	Description
EMCC Geolocation Filter	Choose the geolocation filter that you have configured for use EMCC. Based on the information in the geolocation that associates with a phone that is logged in through Extension Mobility from another cluster, as well as the selected EMCC geolocation filter, Cisco Unified Communications Manager places the phone into a roaming device pool. Cisco Unified Communications Manager determines which roaming device pool to use by evaluating which device pool best matches the phone geolocation information after the EMCC geolocation filter is applied.
EMCC Region Max Audio Bit Rate	This parameter specifies the maximum audio bit rate for all EMCC calls, regardless of the region associated with the other party. The default value is 8 kbps (G.729). **Note** All participating EMCC clusters must specify the same value for the EMCC region max audio bit rate.
EMCC Region Max Video Call Bit Rate (Includes Audio)	This parameter specifies the maximum video call bit rate for all EMCC video calls, regardless of the maximum video call bit rate of the region associated with the other party. The default value is 384. Valid values range from 0 to 8128. **Note** All participating EMCC clusters must specify the same value for the EMCC region max video call bit rate.

EMCC Parameter	Description
EMCC Region Link Loss Type	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 Internet Speech Audio Codec (iSAC) when the link loss type is set as low loss, whereas iSAC is preferred over G.722 when the link loss type is set as lossy. The default value is **Low Loss**.
RSVP SIP Trunk KeepAlive Timer	Specify the number of seconds that Unified Communications Manager waits between sending or receiving KeepAlive messages or acknowledgments between two clusters over EMCC RSVP SIP trunks. 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, Unified Communications Manager releases the RSVP resources with the remote cluster. The default value is 15 seconds. Valid values range from 1second to 600 seconds.
Default Server For Remote Cluster Update	Choose the default server name or IP address of the primary node in this local cluster that has the Cisco Extension Mobility service activated. The remote cluster accesses this node to get information about this local cluster.
Backup Server for Remote Cluster Update	Choose the default server name or IP address of the secondary node in this local cluster that has the Cisco Extension Mobility service activated. The remote cluster accesses this node when the primary node is down to retrieve information about this local cluster.

EMCC Parameter	Description
Remote Cluster Update Interval	Specify an interval, in minutes, during which the Cisco Extension Mobility service on the local node collects information about the remote EMCC cluster. Collected information includes such details as the remote cluster Unified Communications Manager version and service information. The default value is 30. Valid values range from 15 minutes to 10,080 minutes.

Configure Intercluster SIP Trunk for Extension Mobility Cross Cluster

Configure trunks to process inbound or outbound traffic for intercluster PSTN access and RSVP agent services. You can configure one trunk for both PSTN access and RSVP agent services or one trunk for each service. You do not need more than two SIP trunks for extension mobility cross cluster.

Procedure

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

Step 2 Click **Add New**.

Step 3 From the **Trunk Type** drop-down list, choose **SIP Trunk**.

Step 4 From the **Trunk Service Type** drop-down list, choose **Extension Mobility Cross Clusters**.

Step 5 Click **Next**.

Step 6 Configure the fields in the **Trunk Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 7 Click **Save**.

Configure an Intercluster Service Profile for Extension Mobility Cross Cluster

Configure the intercluster service profile to activate extension mobility cross cluster. The profile collects all the configuration that precedes and provides a results report.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Advance Features** > **EMCC** > **EMCC Intercluster Service Profile**.

Step 2 Configure the fields on the **EMCC Intercluster Service Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 3 If no failure messages appear in the popup window, click **Save**.

Configure Remote Cluster Services

Configure the remote cluster for extension mobility cross cluster. This step completes the link between the home cluster with remote (visiting) cluster.

Procedure

Step 1 From 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:

- Click the remote cluster name and verify the fields if the remote cluster that you want to configure appears.
- Click **Add New** if the remote cluster that you want to configure does not appear and configure the following fields:

  a. For the **Cluster Id** field, ensure that the ID matches the enterprise parameter value of the cluster ID of the other clusters.

  b. n 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.

  c. Click **Save**.

Note
For extension mobility cross cluster, the **TFTP** check box should always be disabled.

Extension Mobility Cross Cluster Interactions

Table 37: Extension Mobility Cross Cluster Interactions

Feature	Interaction
Audio	The default maximum audio bit-rate for EMCC login device is set to 8 kbps (G.729).
Call Admission Control (CAC)	• The home cluster is unaware of visiting cluster locations and regions. • The system cannot apply Cisco Unified Communications Manager locations and regions across the cluster boundaries. • RSVP agent-based CAC uses RSVP agents in the visiting cluster.
Call Forwarding	EMCC supports call forwarding.
Cisco Extension Mobility login and logout	User authentication takes place across clusters.

Feature	Interaction
Media resources for the visiting phone	Examples include RSVP agent, TRP, music on mold (MOH), MTP, transcoder, and conference bridge. Media resources are local to the visiting phone (other than RSVP agents).
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 EMCC 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.

Extension Mobility Cross Cluster Restrictions

Table 38: Extension Mobility Cross Cluster Restrictions

Restriction	Description
Unsupported Features	• EMCC does not support the intercom feature, because intercom configuration requires a static device. • EMCC does not support Location CAC, but supports RSVP-based CAC. • Phones with IPv6 only addressing mode do not support EMCC. You can use phones that either use IPv4 or dual stack addressing modes.
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 fails due to the duplicate device error (37). Therefore, for cluster that is 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.

Restriction	Description
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 because of a reset or a logout from the home cluster.
Visiting Device Login Limitations	The extension mobility service in participating clusters performs a periodic remote cluster update. The **Remote Cluster Update Interval** feature parameter controls the update interval. The default interval is 30 minutes. If the extension mobility service on cluster A does not receive a reply from a remote cluster (such as cluster B) for this update, the Remote Cluster window for cluster A shows that the Remote Activated service is set to False for cluster B. In this case, the visiting cluster does not receive any response from the home cluster and sets the Remote Activated values for the home cluster to False. During this interval, a visiting phone may not be able to log in by using EMCC. The visiting phone receives the “Login is unavailable” error message. At this point, a login attempt to EMCC from a visiting phone can fail; the phone displays the “Login is unavailable” error message. This error occurs because the visiting cluster has not yet detected the change of the home cluster from out-of-service to in-service. Remote cluster status change is based on the value of the Remote Cluster Update Interval EMCC feature parameter and on when the visiting extension mobility service performed the last query or update. You can select **Update Remote Cluster Now** in the **Remote Cluster Service Configuration** window (**Advanced Features** > **EMCC** > **EMCC Remote Cluster**) to change Remote Activate values to True, which also allows EMCC logins. Otherwise, after the next periodic update cycle, EMCC logins by visiting phones will return to normal.

EMCC Login Result for Different Cluster Versions with loginType

The following table shows the login result of the Extension Mobility Cross Cluster feature for different cluster versions when the `loginType` parameter is used in the service URL.

Table 39: EMCC Login Result for Different Cluster Versions with loginType

Visiting Cluster Version	Home Cluster Version	`loginType` in Visiting Cluster EM URL*	EMCC Login Result
12.0	12.0	Not mentioned (Default URL)	Success
12.0	12.0	UID, SP, or DN	Success
12.0	11.5 and below	Not mentioned (Default URL)	Success
12.0	11.5 and below	UID, SP, or DN	Fail Fail with error code - 1 **
11.5 and below	12.0	Not mentioned (Default URL)	Success
11.5 and below	12.0	UID, SP, or DN ***	Success

Note

- * Following are the `loginType` parameter options:
  - UID—Users login using User ID and PIN
  - SP—Users login using Self-Service User ID and PIN
  - DN—Users login using Primary Extension and PIN
- ** Fail with error code - 1 — (When EMService could not parse the XML request from EMApp/EMService)
- *** `loginType` will be ignored and the User ID / PIN login prompt gets populated on the phone.

Extension Mobility Cross Cluster and Security Mode for Different Cluster Versions

Note Phone configuration files can be encrypted only if both the home cluster and visiting cluster versions are 9.x or later, and when the TFTP encryption configuration flag is enabled.

During EMCC login, if both the visiting cluster and home cluster versions are in 9.x or later, the phone will behave in various modes as shown in the following table.

Table 40: Supported Security Modes When Both Visiting Cluster and Home Cluster Are In 9.x or later Versions

Home Cluster Version	Home Cluster Mode	Visiting Cluster Version	Visiting Cluster Mode	Visiting Phone Mode	EMCC Status
9.x or later	Mixed	9.x or later	Mixed	Secure	Secure EMCC
9.x or later	Mixed	9.x or later	Mixed	Non-secure	Non-secure EMCC
9.x or later	Mixed	9.x or later	Non-secure	Non-secure	Non-secure EMCC
9.x or later	Non-secure	9.x or later	Mixed	Secure	Login fails
9.x or later	Non-secure	9.x or later	Non-secure	Non-secure	Non-secure EMCC

During EMCC login, if the visiting cluster version is 8.x and the home cluster version is 9.x or later, the phone will behave in various modes as shown in the following table.

Table 41: Supported Security Modes When Visiting Cluster Is In 8.x and Home Cluster Is In 9.x or later Version

Home Cluster Version	Home Cluster Mode	Visiting Cluster Version	Visiting Cluster Mode	Visiting Phone Mode	EMCC Status
9.x or later	Mixed	8.x	Mixed	Secure	Not supported
9.x or later	Mixed	8.x	Mixed	Non-secure	Non-secure EMCC
9.x or later	Mixed	8.x	Non-secure	Non-secure	Non-secure EMCC
9.x or later	Non-secure	8.x	Mixed	Secure	Not supported
9.x or later	Non-secure	8.x	Non-secure	Non-secure	Non-secure EMCC

During EMCC login, if the visiting cluster version is 9.x or later and the home cluster version is 8.x, the phone will behave in various modes as shown in the following table.

Table 42: Supported Security Modes When Visiting Cluster Is In 9.x or later and Home Cluster Is In 8.x Version

Home Cluster Version	Home Cluster Mode	Visiting Cluster Version	Visiting Cluster Mode	Visiting Phone Mode	EMCC Status
8.x	Mixed	9.x or later	Mixed	Secure	Login fails
8.x	Mixed	9.x or later	Mixed	Non-secure	Non-secure EMCC
8.x	Mixed	9.x or later	Non-secure	Non-secure	Non-secure EMCC
8.x	Non-secure	9.x or later	Mixed	Secure	Login fails
8.x	Non-secure	9.x or later	Non-secure	Secure	Non-secure EMCC

Extension Mobility Cross Cluster Troubleshooting

Extension Mobility Application Error Codes

Table 43: Extension Mobility Application Error Codes

Error Code	Phone Display	Quick Description	Reason
201	Please try to login again (201)	Authentication Error	If the user is an EMCC user, this erro occur if "EMCC" is not activated on **Intercluster Service Profile** window
202	Please try to login again (202)	Blank userid or pin	The user enters a blank user ID or PI
204	Login is unavailable (204)	Directory server error	The EMApp sends this error to the pho IMS could not authenticate the user w given PIN.
205	Login is unavailable (205) Logout is unavailable (205)	User Profile Absent	Occurs when the user profile informa cannot be retrieved from the cache or database.
207	Login is unavailable(207) Logout is unavailable(207)	Device Name Empty	Occurs when the device or name tag is in the request URI. This cannot happe real devices and can occur only if the is sent from third-party applications.

Error Code	Phone Display	Quick Description	Reason
208	Login is unavailable(208) Logout is unavailable(208)	EMService Connection Error	The visiting EMApp cannot conn Visiting EMService. (The service not activated.) The visiting EMService cannot co Home EMService (the WAN is do certificates are not trusted.)
210	Login is unavailable(210) Logout is unavailable(210)	Init Fail-Contact Admin	An error (such as a database conn failure) occurred while initializing The error can occur because of a connect to the database during sta
211	Login is unavailable(211) Logout is unavailable(211)	EMCC Not Activated	Occurs when the PSTN is not acti **Intercluster Service Profile** win visiting cluster.
212	Login is unavailable(212)	Cluster ID is invalid	Occurs when a remote cluster upd sending an incorrect cluster ID to cluster.
213	Login is unavailable(213) Logout is unavailable(213)	Device does not support EMCC	Occurs when a device does not sup
215	LoginType invalid(215)	Login Type is invalid	Occurs when `loginType` is invali allowed values are: • `SP` for Self-service User ID • `DN` for Primary Extension • `UID` for User ID
216	DN has multiple users(216)	DN has multiple users	Occurs when the Extension used f is assigned for multiple users as P

Extension Mobility Service Error Codes

Table 44: Extension Mobility Service Error Codes

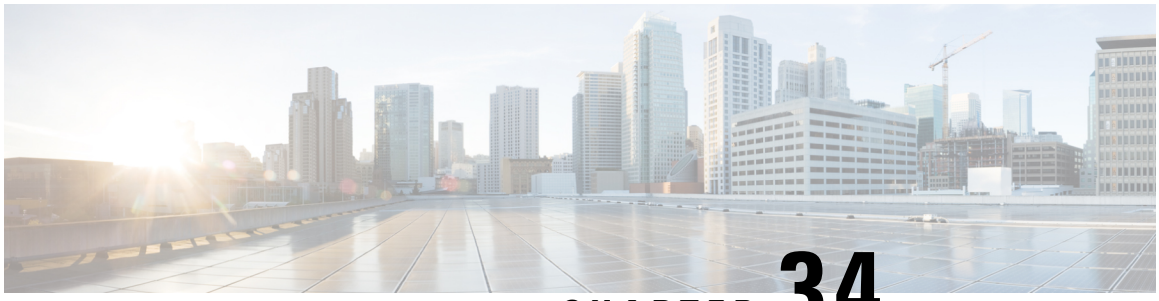
Error Code	Phone Display	Quick Description	Reason
0	Login is unavailable(0) Logout is unavailable(0)	Unknown Error	The EMService failed for an unknow

Error Code	Phone Display	Quick Description	Reason
1	Login is unavailable(1) Logout is unavailable(1)	Error on parsing	When the EMService cannot parse the X request from the EMApp or EMService. T occurs when third-party applications sen incorrect query to login XML (EM API). T can also occur because of a version mism between home and visiting clusters.
2	Login is unavailable(2)	EMCC Authentication Error	The EMCC user credentials cannot be authenticated because the user entered an i PIN.
3	Login is unavailable(3) Logout is unavailable(3)	Invalid App User	Invalid application user. This error comm occurs because of the EM API.
4	Login is unavailable(4) Logout is unavailable(4)	Policy Validation error	The EM Service sends this error when it validate the login or logout request becau unknown reason, an error while querying database or an error while retrieving info from the cache.
5	Login is unavailable(5) Logout is unavailable(5)	Dev. logon disabled	A user logs into a device that has **Enable Extension Mobility** unchecked in the **Ph Configuration** window.
6	Login is unavailable(6) Logout is unavailable(6)	Database Error	Whenever the database returns an excepti executing the query or stored procedure t EM Service requests (login/logout or dev query), the EM Service sends this error c EMApp.
8	Login is unavailable(8) Logout is unavailable(8)	Query type undetermined	No valid query was sent to the EMServic (DeviceUserQuery and UserDeviceQuery ones). This error occurs because of the E or incorrect XML input.
9	Login is unavailable(9) Logout is unavailable(9)	Dir. User Info Error	This error appears in two cases: **1.** IMS returns an exception when it att authenticate a user. **2.** When information about a user canno retrieved either from the cache or dat
10	Login is unavailable(10) Logout is unavailable(10)	User lacks app proxy rights	The user tries to log in on behalf of anoth By default, a CCMSysUser has administ rights.
11	Login is unavailable(11) Logout is unavailable(11)	Device Does not exist	The phone record entry is absent in the d table.

Error Code	Phone Display	Quick Description	Reason
12	Phone record entry is absent in the device table	Dev. Profile not found	No device profile is associated with t user.
18	Login is unavailable(18)	Another user logged in	Another user is already logged in on
19	Logout is unavailable(19)	No user logged in	The system attempted to log out a use not logged in. This error occurs when logout requests from third-party applic API).
20	Login is unavailable(20) Logout is unavailable(20)	Hoteling flag error	**Enable Extension Mobility** is unche **Phone Configuration** window.
21	Login is unavailable(21) Logout is unavailable(21)	Hoteling Status error	The current user status was not retrie either the local cache or database.
22	Login is unavailable(22)	Dev. logon disabled	Occurs when EM is not enabled on de request is sent via EM API or when t button is pressed on phone.
23	Login is Unavailable (23) Logout is Unavailable (23)	User does not exist	Occurs when the given user ID is not any of the remote clusters).
25	Multi-Login Not Allowed (25)	User logged in elsewhere	The user is currently logged in to son phone either in the local cluster or ren
26	Login is unavailable(26) Logout is unavailable(26)	Busy, please try again	Occurs when the EMService has curre the threshold level of Maximum Con Requests service parameter.
28	Login is unavailable(28) Logout is unavailable(28)	Untrusted IP Error	Occurs when the Validate IP Address parameter is set to **True** and the user in or log out from a machine whose I not trusted. For example, a third-party or EM API from a machine is not list Trusted List of Ips service parameter.
29	Login is unavailable(29) Logout is unavailable(29)	ris down-contact admin	The Real-Time Information Server Da (RISDC) cache is not created or initia the EMService is unable to connect t
30	Login is unavailable(30) Logout is unavailable(30)	Proxy not allowed	When login and logout occur through p is set in HTTP header) and the Allow service parameter is set to **False**.
31	Login is unavailable(31) Logout is unavailable(31)	EMCC Not Activated for the user	Occurs when the **Enable Extension M Cross Cluster** check box is not check **End User Configuration** window of cluster.

Error Code	Phone Display	Quick Description	Reason
32	Login is unavailable(32) Logout is unavailable(32)	Device does not support EMCC	Occurs when a device model does not have capability.
33	Login is unavailable(33) Logout is unavailable(33)	No free EMCC dummy device	Occurs when all the EMCC dummy devi in use by other EMCC logins.
35	Login is unavailable(35) Logout is unavailable(35)	Visiting Cluster Information is not present in Home Cluster	Occurs when the home cluster does not h entry for this visiting cluster.
36	Login is unavailable(36) Logout is unavailable(36)	No Remote Cluster	Occurs when the administrator has not ad remote cluster.
37	Login is Unavailable (37) Logout is Unavailable (37)	Duplicate Device Name	Occurs when the same device name exists the home cluster and visiting cluster.
38	Login is unavailable(38) Logout is unavailable(38)	EMCC Not Allowed	Occurs when the home cluster does not v allow EMCC login (The **Enable Extensi Mobility Cross Cluster** check box is not in the home cluster).
39	Please try to login again (201)	Configuration Issue	Occurs when the **Default TFTP Server** **Backup TFTP Server** for EMCC login are not set properly in EMCC Feature Configuration Page. **Note** This is internal error code.
40	Please try to login again (23)	No Response from Remote Host	Occurs when response not getting from R Host. **Note** This is internal error code.
41	PIN change is required	PIN change is required	Occurs when admin enables **User Must** **at Next Login** for PIN. In that case user redirected to Change credentials page. **Note** This is internal error code.
42	Login is unavailable(42) Logout is unavailable(42)	Invalid ClusterID	Occurs when the remote cluster ID is not This error can occur during a remote clus update.
43	Login is unavailable(43)	Device Security mode error	The Device Security Profile that is assoc the EMCC device should be set to Nonse its Device Security Mode.

Error Code	Phone Display	Quick Description	Reason
44	Please try to login again (201)	Configuration Issue	Occurs when the cluster ID is not val **Note** This is internal error code.
45	Login is unsuccessful(45)	Remote Cluster version not supported	Occurs during EMCC login when the cluster version is 9.x and is in mixed phone is in secure mode, and the hom version is 8.x.
46	Login is unsuccessful(46)	Remote Cluster security mode not supported	Occurs during EMCC login when the cluster security mode is in mixed mod is in secure mode, and the home clust nonsecure mode.
47	DN has multiple users(47)	DN has multiple users	Occurs during EMCC login when the used for login is assigned for multipl Primary.



CHAPTER 34

Extension Mobility Roaming Across Clusters

Note To deploy Extension Mobility Roaming Across Clusters, you must be running a minimum release of Cisco Unified Communications Manager 12.0(1)SU1.

- Extension Mobility Roaming Across Clusters Overview, on page 455
- System Requirements for Extension Mobility Roaming Across Clusters, on page 456
- Extension Mobility Roaming Across Clusters Login, on page 456
- ILS Interaction, on page 459
- Extension Mobility Roaming Across Clusters Task Flow, on page 459
- Extension Mobility Roaming Across Clusters Interactions and Restrictions, on page 464
- Different Types of Extension Mobility, on page 464
- Extension Mobility Roaming Across Clusters Troubleshooting, on page 465

Extension Mobility Roaming Across Clusters Overview

Extension Mobility Roaming Across Clusters gives users the ability to roam across multiple cluster and make or receive calls even when the user's home cluster is down. This feature leverages the Intercluster Lookup Service (ILS) to replicate Extension Mobility users' directory numbers across all the clusters.

When a user logs in to a roaming cluster, their phone registers to the roaming cluster using the directory number. Unlike Extension Mobility Cross Cluster (EMCC), where the phone from the visiting cluster registers to the home cluster, the roaming feature allows the user to maintain their registration in whichever cluster they are visiting.

Configuration Overview

To deploy this feature, you must do the following:

- Set up an ILS network—ILS is used to synchronize directory number across the clusters.

  For details on configuring ILS, see the Configure Intercluster Lookup Service chapter in the System Configuration Guide for Cisco Unified Communications Manager.

- Set up a uniform dial plan—you require a uniform dial plan across the ILS network.

To set up a dial plan, see Configure the Dial Plan chapter in the System Configuration Guide for Cisco Unified Communications Manager.

- Device profile and user information must be synced in all the clusters.
- Configure Extension Mobility.
- Configure Roaming access to your Extension Mobility users.

System Requirements for Extension Mobility Roaming Across Clusters

The following system requirements exist for Cisco Unified Communications Manager:

- Cisco Unified Communications Manager, Release 12.0(1)SU1 or higher.
- Cisco Extension Mobility service must be running.
- Intercluster Lookup Service must be running.

Extension Mobility Roaming Across Clusters Login

Login Terminology

The following figure depicts the home cluster versus a roaming cluster in Extension Mobility Roaming across Cluster.

Figure 7: Home Cluster vs. Roaming Cluster



Home Cluster
Home cluster is a cluster, where the user configuration such as User Device profile, Dial Plans reside here.

Roaming Cluster
Roaming cluster is a cluster, where users can do the Extension Mobility login to any Extension Mobility capable phone just like in their home cluster.

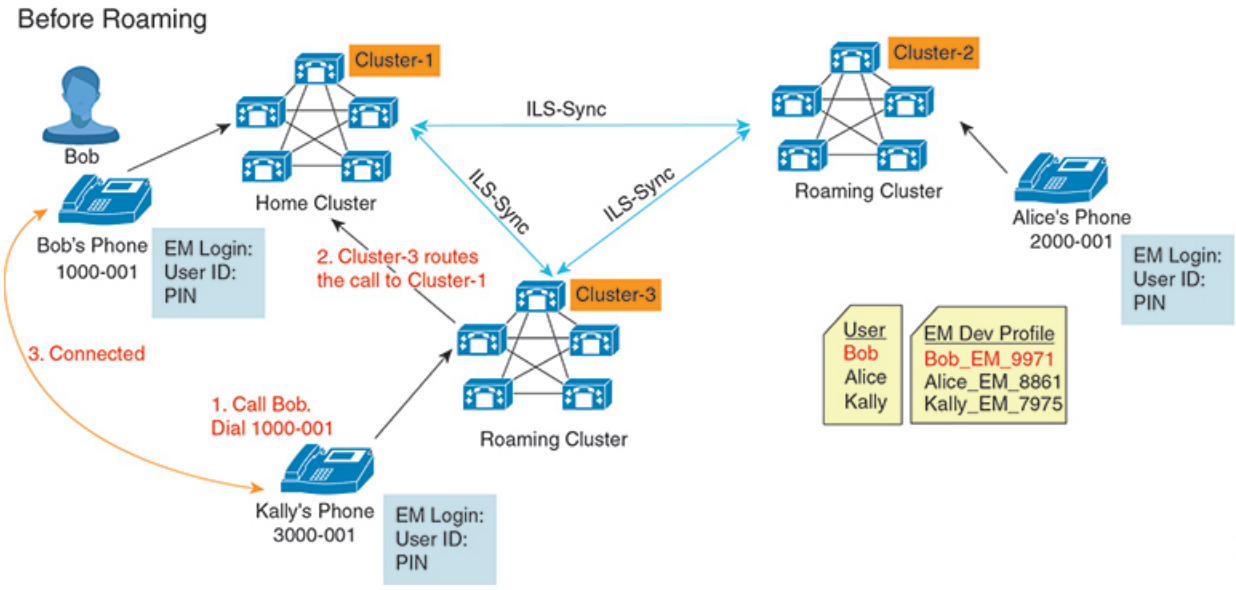
Superuser
A superuser is a user, who is associated to the **Standard EM Roaming Across Clusters Super Users** access control group. This user has a privilege to do the Extension Mobility login from a roaming cluster and can make or receive calls.

Note Superuser information must be shared across all the clusters irrespective of the cluster in which the user is logging-in.

Login Process

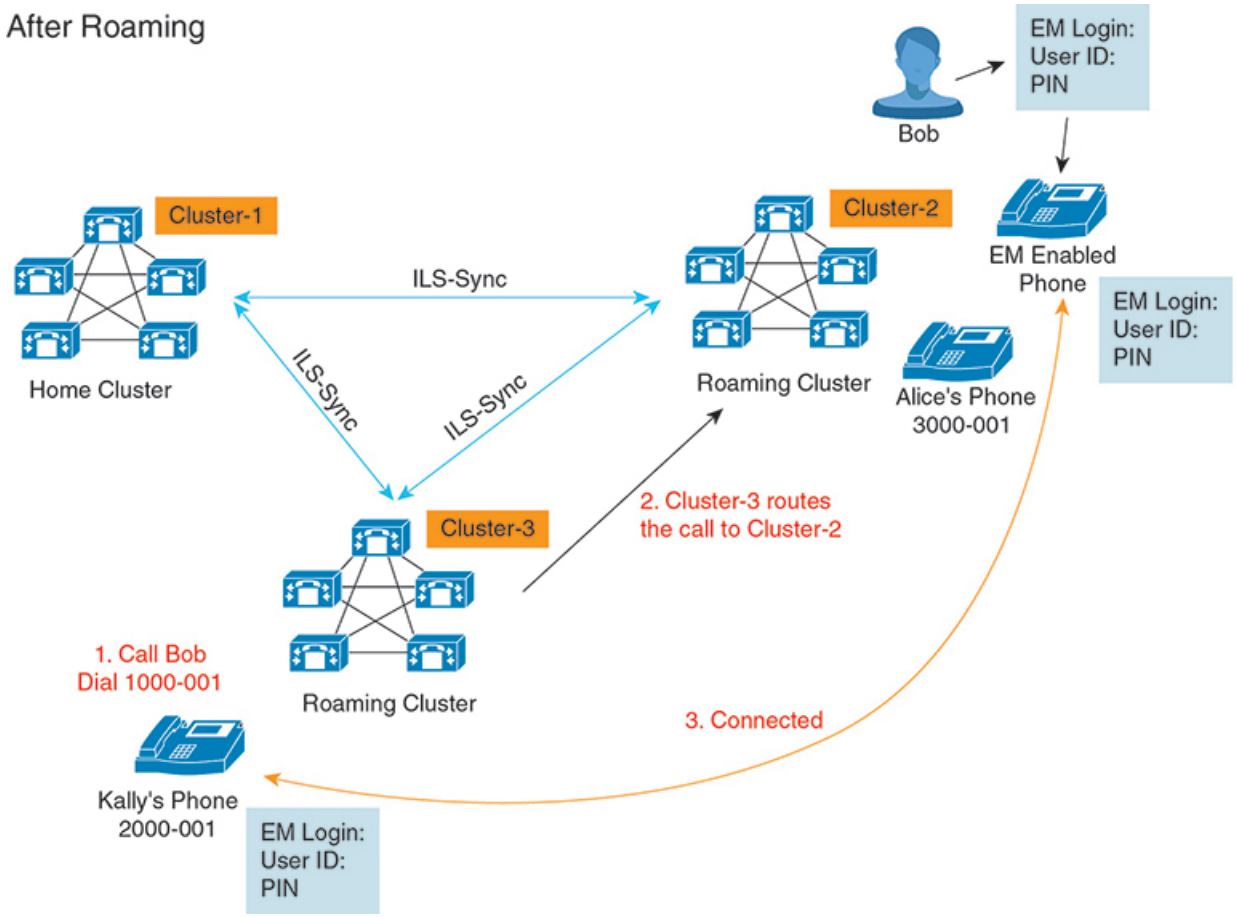
Cisco Unified Communications Manager supports the Extension Mobility login for a superuser created across multiple clusters. Extension Mobility login, in the roaming cluster allows superuser to access their phone settings, such as line appearances, services, dial plans. A superuser can make or receive calls from the roaming cluster, in the same way as they do in the home cluster.

Figure 8: Call Flow When a User Is in Home Cluster



In the preceding figure, let us assume Bob's DN as 1000-001, Alice's DN as 2000-001 and Kally's DN as 3000-001 registered with Cluster 1, 2 and 3 respectively. When Kally dials Bob's DN 1000-001, Cluster 3 routes the call to Cluster 1 and Bob and Kally are connected.

Figure 9: Call Flow When User Is in Roaming Cluster



Let us assume Bob's home cluster is down and Bob is configured as superuser who can roam across the clusters. When Bob moves to Cluster 2 and does Extension Mobility Login, hosting phone gets re-registered with Bob's settings. Once the login is successful, all other clusters are updated with Bob's new location. Now when Kally dials Bob's DN 1000-01, Cluster 3 routes the call to Cluster 2 and Bob and Kally are connected. Similarly, Bob can call Kally by dialing DN 3000-001.

Note

- If a superuser did the Extension Mobility login to another cluster, user will automatically log out from the home cluster. If the cluster is down, it waits until the cluster is up to log out from the user's previous login.
- Extension Mobility Roaming Across Clusters supports the multi login behavior. Hence, superuser can login from multiple devices within the same cluster but not across the clusters.

ILS Interaction

In Cisco Unified CM Administration, you can configure ILS on a pair of clusters and then join those clusters to form an ILS network. Once you have established the ILS network, you can join additional clusters to the network without having to configure the connections between each cluster.

Whenever Extension Mobility login or logout occurs, ILS sync starts to update the available information to other clusters.

Note Configuring user as superuser automatically initiates the ILS sync irrespective of Directory Number configuration for ILS.

For more information, see the Configure Intercluster Lookup Service chapter in the *System Configuration Guide for Cisco Unified Communications Manager* at https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html.

Extension Mobility Roaming Across Clusters Task Flow

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the Extension Mobility feature.
Step 2	Configure Extension Mobility by completing, the following subtasks in order: • Activate Extension Mobility Services, on page 405 • Configure the Cisco Extension Mobility Phone Service, on page 406 • Create an Extension Mobility Device Profile for Users, on page 407 • Associate a Device Profile to a User, on page 407 • Subscribe to Extension Mobility, on page 408	Configure Extension Mobility to allow users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones when they login from remote cluster. Perform this task flow on both home and remote clusters, so that users will be able to access settings from either a home or remote cluster.
Step 3	Configure Roaming for Extension Mobility Users, on page 463	Use this procedure to give Extension Mobility users the ability to roam between different clusters in an ILS network, while using the same login credentials.

Generate a Phone Feature List

Generate a phone feature list report to determine which devices support the feature that you want to configure.

Procedure

Step 1 From Cisco Unified Reporting, choose **System Reports**.

Step 2 From the list of reports, click **Unified CM Phone Feature List**.

Step 3 Perform one of the following steps:

- Choose **Generate New Report** (the bar chart icon) to generate a new report.
- Choose **Unified CM Phone Feature List** if a report exists.

Step 4 From the **Product** drop-down list, choose **All**.

Step 5 Click the name of the feature that you want to configure.

Step 6 Click **Submit**, to generate the report.

Activate Extension Mobility Services

Procedure

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

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

Step 3 Activate the following services:

a) Cisco CallManager
b) Cisco Tftp
c) Cisco Extension Mobility
d) ILS Service

Note
You must choose publisher node to activate the ILS services.

Step 4 Click **Save**.

Step 5 Click **OK**.

Configure the Cisco Extension Mobility Phone Service

Configure the extension mobility IP phone service to which users can later subscribe to access extension mobility.

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.

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 Unified Communications Manager where Cisco Extension Mobility is activated and running.

It should be a IPv4 address.

Example:

`http://123.45.67.89:8080/emapp/EMAppServlet?device=#DEVICENAME#`

Example:

`http://[2001:0001:0001:0067:0000:0000:0000:0134]:8080/emapp/EMAppServlet?device=#DEVICENAME#`

This format allows a user to sign-in using User ID and PIN. You can configure more sign-in options for IP phone users who have subscribed to the extension mobility service. To configure more sign-in options, append the `loginType` parameter to the Service URL, in the following formats:

- `loginType=DN` enables users to sign in using Primary Extension and PIN.

  The Service URL format is:`http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=DN`.

- `loginType=SP` enables users to sign in using Self Service User ID and PIN.

  The Service URL format is:`http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=SP`.

- `loginType=UID` enables users to sign in using User ID and PIN.

  The Service URL format is:`http://<IP Address>:8080/emapp/EMAppServlet?device=#DEVICENAME#&loginType=UID`.

If you do not append `loginType` to the end of the URL, the default sign-in option displayed is User ID and PIN.

Step 5 In the **Service Type** field, choose whether the service is provisioned to the Services, Directories, or Messages button.

Step 6 Click **Save**.

Create an Extension Mobility Device Profile for Users

Configure an extension mobility device profile. This profile acts as a virtual device that maps onto a physical device when a user logs in to extension mobility. The physical device takes on the characteristics in this profile.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings and choose an existing device profile from the resulting list.
- Click **Add New** to add a new device profile and choose an option from the **Device Profile Type**. Click **Next**.
- Choose a device protocol from the **Device Protocol** drop-down list and click **Next**.

Step 3 Configure the fields. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Step 5 From the **Association Information** section, click **Add a new DN**.

Step 6 In the **Directory Number** field, enter the directory number and click **Save**.

Step 7 Click **Reset** and follow the prompts.

Associate a Device Profile to a User

Associate a device profile to users so that they can access their settings from a different phone. You associate a user device profile to a user in the same way that you associate a physical device.

Tip You can use the Bulk Administration Tool (BAT) to add and delete several user device profiles for Cisco Extension Mobility at one time. See the Bulk Administration Guide for Cisco Unified Communications Manager.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing user, enter search criteria, and choosing an existing user from the resulting list.
- Click **Add New** to add a new user.

Step 3 Under **Extension Mobility**, locate the device profile that you created and move it from **Available Profiles** to **Controlled Profiles**.

Step 4 Check the **Home Cluster** check box.

Step 5 Click **Save**.

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.

Procedure

Step 1 Perform one of the following tasks from Cisco Unified CM Administration:

- Choose **Device** > **Phone**, specify search criteria, click **Find**, and choose a phone which users will use for extension mobility.
- Choose **Device** > **Device Settings** > **Device Profile**, specify search criteria, click **Find**, and choose the device profile that you created.

Step 2 From the **Related Links** drop-down list, choose **Subscribe/Unsubscribe Services**, and then 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**.

Step 6 Click **Save** and close the popup window.

Configure Roaming for Extension Mobility Users

Use this procedure to give Extension Mobility users the ability to roam between different clusters in an ILS network, while using the same login credentials. To do this, you must assign the selected users to the **Standard EM Roaming Across Clusters Super Users** access control group.

Before you begin

An ILS network must have been set up as ILS is used to replicate user and login information across the clusters.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **User Settings** > **Access Control Group**.

Step 2 Click **Find** and select the **Standard EM Roaming Across Clusters Super Users** group.

Step 3 Click **Add End Users to Group** button. The **Find and List Users** pop-up window appears.

Step 4 Click **Find** and select all the users to whom you want to provide roaming ability.

Step 5 Click **Add Selected**.

Extension Mobility Roaming Across Clusters Interactions and Restrictions

Extension Mobility Roaming Across Clusters Interactions

This section lists the interactions of the Extension Mobility Roaming across Cluster with other Cisco Unified Communications Manager Administration components.

- Extension Mobility
- Inter-Cluster Lookup Service (ILS)

Extension Mobility Roaming Across Clusters Restrictions

This section lists the restrictions of the Extension Mobility Roaming across Cluster with other Cisco Unified Communications Manager Administration components.

- If the hub ILS is down, the spokes connected to it will not synchronize until the hub is back.

Different Types of Extension Mobility

The following table lists the different types of Extension Mobility features available in Cisco Unified Communications Manager and their differences.

Table 45: Differences Between EM, EMCC, and Extension Mobility Roaming Across Cluster

	Extension Mobility (EM)	Extension Mobility Cross Cluster (EMCC)	Extension Mobility Roaming Across Cluster
Description	Allows users to temporarily access their phone settings from other phones in the same cluster.	Allows users to access their phone settings from a phone in another cluster.	Allows user to roam across other clusters using own login credentials.
When the user logs in to a phone in another cluster	N/A	The remote cluster phone registers to the user's home cluster, accessing the settings in the home cluster.	The roaming cluster phone registers in the roaming cluster only.
Intercluster	Single cluster only	Multiple clusters	Multiple clusters
Configuration	Single cluster only	EMCC must be configured in the home cluster and each cluster that the user visit.	Extension Mobility Roaming must be configured in all the cluster.

	Extension Mobility (EM)	Extension Mobility Cross Cluster (EMCC)	Extension Mobility Roaming Across Cluster
User Information	Single cluster only	Must be maintained in all clusters.	Superuser information maintained in all the cluster.

Extension Mobility Roaming Across Clusters Troubleshooting

This section provides information about error codes for EMApp and EMService.

Authentication Error

Problem "Error 201 Authentication Error" appears on the phone.

Solution The user should check that the correct user ID and PIN were entered; the user should check with the system administrator that the user ID and PIN are correct.

Blank User ID or PIN

Problem "Error 202 Blank User ID or PIN" appears on the phone.

Solution Enter a valid user ID and PIN.

Busy Please Try Again

Problem "Error 26 Busy Please Try Again" appears on the phone.

Solution Check whether the number of concurrent login and logout requests is greater than the **Maximum Concurrent requests** service parameter. If so, lower the number of concurrent requests.

Note To verify the number of concurrent login and logout requests, use the Cisco Unified Real-Time Monitoring Tool to view the Requests In Progress counter in the Extension Mobility object. For more information, see the Cisco Unified Real-Time Monitoring Tool Administration Guide at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html

Database Error

Problem "Error 6 Database Error" appears on the phone.

Solution Check whether a large number of requests exists. If a large number of requests exists, the Requests In Progress counter in the Extension Mobility object counter shows a high value. If the requests are rejected because of a large number of concurrent requests, the Requests Throttled counter also shows a high value. Collect detailed database logs.

Dev Logon Disabled

Problem "Error 22 Dev Logon Disabled" appears on the phone.

Solution Verify that you checked the **Enable Extension Mobility** check box in the **Phone Configuration** window (**Device** > **Phone**).

Device Name Empty

Problem "Error 207 Device Name Empty" appears on the phone.

Solution Check that the URL that is configured for Cisco Extension Mobility is correct. See the Related Topics section for more information.

Related Topics

Configure the Cisco Extension Mobility Phone Service, on page 406

EM Service Connection Error

Problem "Error 207 EM Service Connection Error" appears on the phone.

Solution Verify that the Cisco Extension Mobility service is running by selecting **Tools** > **Control Center—Feature** in Cisco Unified Serviceability.

Host Not Found

Problem The "Host Not Found" error message appears on the phone.

Solution Check that the Cisco Tomcat service is running by selecting **Tools** > **Control Center—Network Services** in CIsco Unified Serviceability.

HTTP Error

Problem HTTP Error (503) appears on the phone.

Solution

- If you get this error when you press the **Services** button, check that the Cisco IP Phone Services service is running by selecting **Tools** > **Control Center—Network Services** in Cisco Unified Serviceability.
- If you get this error when you select Extension Mobility service, check that the Cisco Extension Mobility Application service is running by selecting **Tools** > **Control Center—Network Services** in Cisco Unified Serviceability.

Phone Resets

Problem After users log in or log out, their phones reset instead of restarting.

Possible Cause Locale change is the probable cause of the reset.

Solution No action is required. If the user locale that is associated with the logged-in user or profile is not the same as the locale or device, after a successful login the phone will restart and then reset. This pattern occurs because the phone configuration file is rebuilt.

Phone Services Unavailable After Login

Problem After logging in, the user finds that the phone services are not available.

Possible Cause This problem occurs because the user profile had no services associated with it when it was loaded on the phone.

Solution

- Ensure that the user profile includes the Cisco Extension Mobility service.
- Change the configuration of the phone where the user is logged in to include Cisco Extension Mobility. After the phone is updated, the user can access the phone services.

Phone Services Unavailable After Logout

Problem After a user logs out and the phone reverts to the default device profile, the phone services are no longer available.

Solution Subscribe the phone to the Cisco Extension Mobility service.

User Logged in Elsewhere

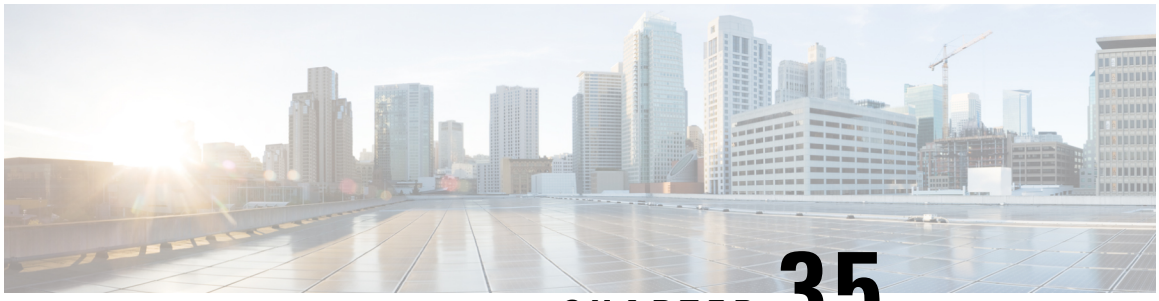
Problem "Error 25 User Logged in Elsewhere" appears on the phone.

Solution Check whether the user is logged in to another phone. If multiple logins must be allowed, ensure that the **Multiple Login Behavior** service parameter is set to **Multiple Logins Allowed**.

User Profile Absent

Problem "Error 205 User Profile Absent" appears on the phone.

Solution Associate a device profile to the user.



CHAPTER 35

Hold Reversion

- Hold Reversion Overview, on page 469
- Hold Reversion Prerequisites, on page 470
- Hold Reversion Configuration Task Flow, on page 470
- Hold Reversion Interactions, on page 473
- Hold Reversion Restrictions, on page 474

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

Note 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.
- Review Hold Reversion Prerequisites, on page 470

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Run a phone feature list report to determine which phones support the Hold Reversion feature.
Step 2	Configure Call Focus Priority for Hold Reversion, on page 471	Configure the call focus priority setting against the device pool for your phones.
Step 3	Perform one of the following procedures: • Configure Hold Reversion Timer Defaults for Cluster, on page 471 • Configure Hold Reversion Timer Settings for Phone, on page 472	Configure the Hold Reversion timer settings. You can configure the timer using a clusterwide service parameter, or configure the settings on an individual phone line. **Note** The settings on an individual phone line override the clusterwide service parameter settings.

Configure Call Focus Priority for Hold Reversion

As an administrator, you can prioritize incoming calls and reverted calls. By default, all incoming calls are handled before reverted calls, however you can change the call focus priority so that reverted calls take precedence.

Before you begin

Generate a Phone Feature List, on page 5

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 471
- Configure Hold Reversion Timer Settings for Phone, on page 472

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 471

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 **CallManager** service.

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

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 471 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

Table 46: Hold Reversion Feature Interactions

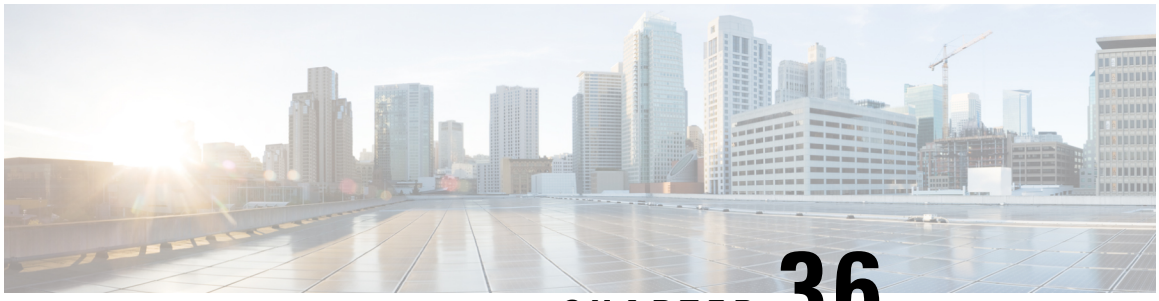
Feature	Interactions
Music on Hold	MOH is supported on a reverted call if MOH is configured for a normal held call.
Call Park	If hold reversion is invoked and the held party presses the **Park** softkey, the holding party still receives hold reversion alerts and can retrieve the call. When the holding 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.
MLPP	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.
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*

Feature	Interactions
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

Feature	Restriction
Cisco Extension Mobility and Cisco Web Dialer	Cisco Extension Mobility and Cisco Web Dialer features do not support the hold reversion feature.
SCCP phones	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.
Directory numbers	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.
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 this feature, hold reversion gets disabled on all other devices that share this line.
Ring settings	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. **Note** When an IP Phone call is on normal hold, the ring settings (Phone Idle) from the Call Manager is applied.

Feature	Restriction
Maximum number of reverted calls	The maximum number of reverted calls on a line equals the maximum number of calls on your system.
CTI Applications	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: • *Cisco Unified TAPI Developers Guide for Cisco Unified Communications Manager* • *Cisco Unified JTAPI Developers Guide for Cisco Unified Communications Manager*
Cisco Unified IP Phones	You cannot configure hold reversion settings for DNs that are associated with phones that do not support this feature. Only Cisco Unified IP Phones that support the hold reversion feature display the hold reversion timer settings in the Directory Number Configuration window. When Hold Reversion is configured for the system, the phone must support the feature or the feature does not activate. See Cisco Unified IP Phone administration guides for Cisco Unified IP Phone models that support hold reversion and this version of Unified Communications Manager for any phone restrictions with hold reversion.



CHAPTER 36

Accessing Hunt Groups

- Hunt Group Overview, on page 477
- Hunt Group Prerequisites, on page 478
- Hunt Group Configuration Task Flow, on page 478
- Hunt Group Interactions, on page 483
- Hunt Group Restrictions, on page 484

Hunt Group Overview

A Hunt Group is a group of lines that are organized hierarchically, so that if the first number in the hunt group list is busy, the system dials the second number. If the second number is busy, the system dials the next number, and so on.

The phone users can log in to or log out of the hunt groups by using the HLog softkey or the Hunt Group line button on the IP phone. The phone provides a visual status of the login state, so that the user can determine whether they are logged in to one or more of their line groups.

The Hunt Group feature provides the following functions:

- The HLog softkey on the IP phone allows the user to toggle between login and logout of phone.
- A hunt group allows a caller to automatically find an available line from amongst a group of extensions.
- The Hunt Group Log Off feature allows phone users to prevent their phones from receiving incoming calls that get routed to directory numbers. Regardless of the phone status, the phone rings normally for incoming calls that are not calls to one or more line groups associated with the phone.

Note The directory numbers (DNs) belong to line groups that are associated with the phone.

- System administrators can log in or log out the users from the phones that are automatically logged into hunt groups.
- The HLog softkey allows a phone user to log a phone out of all line groups to which the phone directory numbers belong.

- From Cisco Unified Communications Manager Release 9.0 onward, the Hunt Group Log Off feature enables the use of mobile device as a desk phone. When you use the Hlog softkey through your mobile client, you no longer receive calls that are placed to the hunt pilot.

Hunt Group Prerequisites

- The phones must be running Skinny Client Control Protocol (SCCP) or Session Initiation Protocol (SIP).
- The phone ringtone file must be located in the TFTP directory (/usr/local/cm/tftp).

Hunt Group Configuration Task Flow

Before you begin

- Review Hunt Group Prerequisites, on page 478

Procedure

	Command or Action	Purpose
Step 1	Configure a Softkey Template for Hunt Group, on page 479	Configure a softkey template for the HLog softkey.
Step 2	To Associate a Softkey Template with a Common Device Configuration, on page 480, complete the following subtasks: • Add a Softkey Template to a Common Device Configuration, on page 480 • Associate a Common Device Configuration with a Phone, on page 481	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 481	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 482	Configure phones to automatically log in to or log out of hunt groups and hunt lists.

Configure a Softkey Template for Hunt Group

The HLog softkey appears on the phone when the phone is in the following call states:

- Connected
- On Hook
- Off Hook

Note You must create a new softkey template to configure the HLog softkey. You cannot configure the HLog softkey in a standard softkey template.

Use this procedure to make the HLog softkey available:

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.

- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one of the following procedures:

- Add a Softkey Template to a Common Device Configuration, on page 480
- Associate a Softkey Template with a Phone, on page 481

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 481.

Before you begin

Configure a Softkey Template for Hunt Group, on page 479

Procedure

	Command or Action	Purpose
Step 1	Add a Softkey Template to a Common Device Configuration, on page 480	
Step 2	Associate a Common Device Configuration with a Phone, on page 481	

Add a Softkey Template to a Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.
a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Before you begin

Add a Softkey Template to a Common Device Configuration, on page 480

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey Template with a Phone

This procedure is optional. You can use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration: use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Before you begin

Configure a Softkey Template for Hunt Group, on page 479

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**.
A dialog box appears with a message to press **Reset** to update the phone settings.

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 Administration Guide for Cisco Unified Communications Manager for information on hunt groups and hunt lists.

Procedure

Step 1 From 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**.

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.

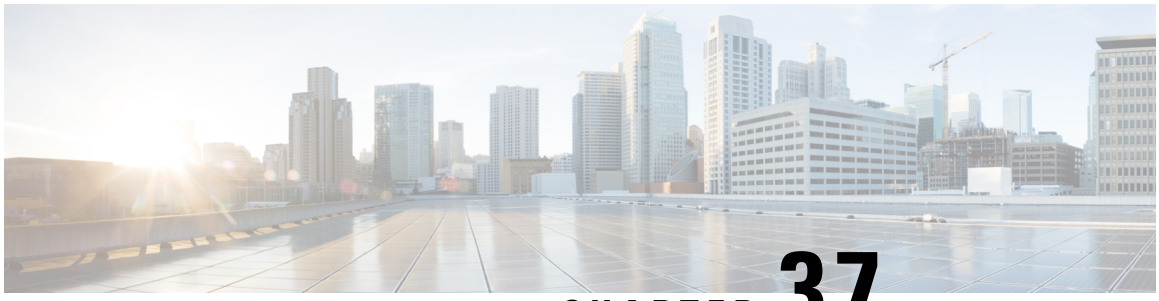
Step 5 Click **Save**.
The window refreshes, and Cisco Unified Communications Manager updates the service parameter with your changes.

Hunt Group Interactions

Feature	Interaction
Non-shared-line Directory Number	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.
Shared-line Directory Number	Because the Log Out of Hunt Group feature is device-based, when a user logs a phone out, the feature affects only the logged-out phone. Calls to a line group that contains a shared-line directory number behave as follows: • The DN does not ring if all phones that share that DN are logged out. • The DN does ring if one or more phones that share the DN are logged in. • The audible ring on a phone that is logged out is turned off by default. Cisco Unified Communications Manager provides a system parameter that can be set, so that a different ring tone plays when a call comes in to a logged-out hunt group member.

Hunt Group Restrictions

Restriction	Description
Multiple Line Groups	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.
7940, 7960, and third-party SIP phones	• When a phone that is running SIP (7906, 7911, 7941, 7961,) is logged in to hunt groups and Call Forward All is activated, the call gets presented to the phone that is running SIP. • When 7940 and 7960 phones that are running SIP are logged in to hunt groups and Call Forward All is activated, the phones get skipped and the next phone in the line group rings. • 7940 and 7960 phones that are running SIP and third-party phones that are running SIP can be logged in to or logged out of hunt groups by using the **Phone Configuration** window, but no softkey support exists. • 7940 and 7960 phones that are running SIP and third-party phones that are running SIP do not show “Logged out of hunt groups” on the status line. • 7940 and 7960 phones that are running SIP and third-party phones that are running SIP do not play the Hunt Group Logoff Notification tone regardless of whether the tone is configured.



CHAPTER 37

Malicious Call Identification

- Malicious Call Identification Overview, on page 485
- Malicious Call Identification Prerequisites, on page 485
- Malicious Call Identification Configuration Task Flow, on page 486
- Malicious Call Identification Interactions, on page 492
- Malicious Call Identification Restrictions, on page 493
- Malicious Call ID Troubleshooting, on page 494

Malicious Call Identification Overview

You can configure the Malicious Call Identification (MCID) feature to track troublesome or threatening calls. Users can report these calls by requesting that Cisco Unified Communications Manager identify and register the source of the incoming call in the network.

When the MCID feature is configured, the following actions take place:

1. The user receives a threatening call and presses Malicious call (or enters the feature code *39 if using a POTS phone that is connected to an SCCP gateway).
2. Cisco Unified Communications Manager sends the user a confirmation tone and a text message, if the phone has a display, to acknowledge receiving the MCID notification.
3. Cisco Unified Communications Manager updates the call details record (CDR) for the call with an indication that the call is registered as a malicious call.
4. Cisco Unified Communications Manager generates the alarm and local syslogs entry that contains the event information.
5. Cisco Unified Communications Manager sends an MCID invocation through the facility message to the connected network. The facility information element (IE) encodes the MCID invocation.
6. After receiving this notification, the PSTN or other connected network can take actions, such as providing legal authorities with the call information.

Malicious Call Identification Prerequisites

- Gateways and connections that support MCID:
  - PRI gateways that use the MGCP PRI backhaul interface for T1 (NI2) and E1 (ETSI) connections
  - H.323 trunks and gateways

- IP Phones that support MCID

Malicious Call Identification Configuration Task Flow

Before you begin

- Review Malicious Call Identification Prerequisites, on page 485

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the MCID feature.
Step 2	Set Malicious Call ID Service Parameter, on page 487	Enable Cisco Unified Communications Manager to flag a call detail record (CDR) with the MCID indicator.
Step 3	Configure Malicious Call ID Alarms, on page 487	Configure alarms to ensure that alarm information displays in the system logs.
Step 4	Configure a Softkey Template for Malicious Call Identification, on page 488	Configure a softkey template with MCID. **Note** The Cisco Unified IP Phones 8900 and 9900 Series support MCID with feature button only.
Step 5	To Associate a Softkey Template with a Common Device Configuration, on page 489, complete the following subtasks: • Add a Softkey Template to a Common Device Configuration, on page 489 • Associate a Common Device Configuration with a Phone, on page 490	Optional. To make the softkey template available to phones, you must complete either this step or the following step. Follow this step if your system uses a **Common Device Configuration** to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.
Step 6	Associate a Softkey Template with a Phone, on page 490	Optional. Use this procedure either as an alternative to associating the softkey template with the Common Device Configuration, or in conjunction with the Common Device Configuration. Use this procedure in conjunction with the Common Device Configuration if you need assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.
Step 7	To Configure Malicious Call Identification Button, on page 491, complete the following subtasks:	Perform this step to add and configure the MCID button to a phone.

	Command or Action	Purpose
	• Configure Malicious Call ID Phone Button Template, on page 491 • Associate a Button Template with a Phone , on page 492	

Set Malicious Call ID Service Parameter

To enable Unified Communications Manager to flag a CDR with the MCID indicator, you must enable the CDR flag.

Before you begin

Configure Malicious Call ID Alarms, on page 487

Procedure

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

Step 2 From the **Server** drop-down list, choose the Unified Communications Manager server name.

Step 3 From the **Service** drop-down list, choose **Cisco CallManager**.
The **Service Parameter Configuration** window displays.

Step 4 In the System area, set the **CDR Enabled Flag** field to **True**.

Step 5 Click **Save**.

Configure Malicious Call ID Alarms

In the Local Syslogs, you must set the alarm event level and activate alarms for MCID.

Cisco Business Edition 5000 systems support only one node.

Before you begin

Set Malicious Call ID Service Parameter, on page 487

Procedure

Step 1 From Cisco Unified Serviceability, choose **Alarm** > **Configuration**.
The **Alarm Configuration** window displays.

Step 2 From the **Server** drop-down list, choose the Unified Communications Manager server and click **Go**.

Step 3 From the **Service Group** drop-down list, choose **CM Services**. The **Alarm Configuration** window updates with configuration fields.

Step 4 From the **Service** drop-down list, choose **Cisco CallManager**.

Step 5 Under Local Syslogs, in the **Alarm Event Level** drop-down list, choose **Informational**.

The **Alarm Configuration** window updates with configuration fields.

Step 6 Under Local Syslogs, check the **Enable Alarm** check box.

Step 7 If you want to enable the alarm for all nodes in the cluster, check the **Apply to All Nodes** check box.

Step 8 To turn on the informational alarm, click **Update**.

Configure a Softkey Template for Malicious Call Identification

Note Skinny Client Control Protocol (SCCP) IP phones use a softkey to invoke the MCID feature.

Before you begin

Configure Malicious Call ID Alarms, on page 487

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 In the **Select a call state to configure** field, choose **Connected**.
The list of Unselected Softkeys changes to display the available softkeys for this call state.

Step 7 In the **Unselected Softkeys** drop-down list, choose **Toggle Malicious Call Trace (MCID)**.

Step 8 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 9 Click **Save.**

Associate a Softkey Template with a Common Device Configuration

Optional. There are two ways to associate a softkey template with a phone:

- Add the softkey template to the **Phone Configuration**.
- Add the softkey template to the **Common Device Configuration**.

The procedures in this section describe how to associate the softkey template with a **Common Device Configuration**. Follow these procedures if your system uses a **Common Device Configuration** to apply configuration options to phones. This is the most commonly used method for making a softkey template available to phones.

To use the alternative method, see Associate a Softkey Template with a Phone, on page 490.

Before you begin

Configure a Softkey Template for Malicious Call Identification, on page 488

Procedure

	Command or Action	Purpose
Step 1	Add a Softkey Template to a Common Device Configuration, on page 489	
Step 2	Associate a Common Device Configuration with a Phone, on page 490	

Add a Softkey Template to a Common Device Configuration

Before you begin

Configure a Softkey Template for Malicious Call Identification, on page 488

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Before you begin

Add a Softkey Template to a Common Device Configuration, on page 489

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate a Softkey Template with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Configure Malicious Call Identification Button

The procedures in this section describe how to configure the Malicious Call Identification button.

Before you begin

Configure Malicious Call ID Alarms, on page 487

Procedure

	Command or Action	Purpose
Step 1	Configure Malicious Call ID Phone Button Template, on page 491.	Perform this step to assign Malicious Call Identification button features to line or speed dial keys.
Step 2	Associate a Button Template with a Phone , on page 492	Perform this step to configure the Malicious Call Identification button for a phone.

Configure Malicious Call ID Phone Button Template

Before you begin

Configure Malicious Call ID Alarms, on page 487

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate a Button Template with a Phone

Before you begin

Configure Malicious Call ID Phone Button Template, on page 491

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Malicious Call Identification Interactions

Table 47: Malicious Call Identification Interactions

Feature	Interaction
Conference Calls	When a user is connected to a conference, the user can use the MCID feature to flag the call as a malicious call. Cisco Unified Communications Manager sends the MCID indication to the user, generates the alarm, and updates the CDR. However, Cisco Unified Communications Manager does not send an MCID invoke message to the connected network that might be involved in the conference.
Extension Mobility	Extension Mobility users can have the MCID softkey as part of their user device profile and can use this feature when they are logged on to a phone.
Call Detail Records	To track malicious calls by using CDR, you must set the **CDR Enabled Flag** to **True** in the Cisco CallManager service parameter. When the MCID feature is used during a call, the CDR for the call contains **CallFlag=MALICIOUS** in the Comment field.

Feature	Interaction
Alarms	To record alarms for the MCID feature in the Local Syslogs, you must configure alarms in Cisco Unified Serviceability. Under **Local Syslogs**, enable alarms for the **Informational** alarm event level. When the MCID feature is used during a call, the system logs an SDL trace and a Cisco Unified Communications Manager trace in alarms. You can view the **Alarm Event Log** by using Cisco Unified Serviceability. The traces provide the following information: • Date and time • Type of event: Information • Information: The Malicious Call Identification feature is invoked in Cisco Unified Communications Manager • Called Party Number • Called Device Name • Called Display Name • Calling Party Number • Calling Device Name • Calling Display Name • Application ID • Cluster ID • Node ID For more information about alarms and traces, see the *Cisco Unified Serviceability Administration Guide* at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.
Cisco ATA 186 analog phone ports	The Cisco ATA 186 analog phone ports support MCID by using the feature code (*39).

Malicious Call Identification Restrictions

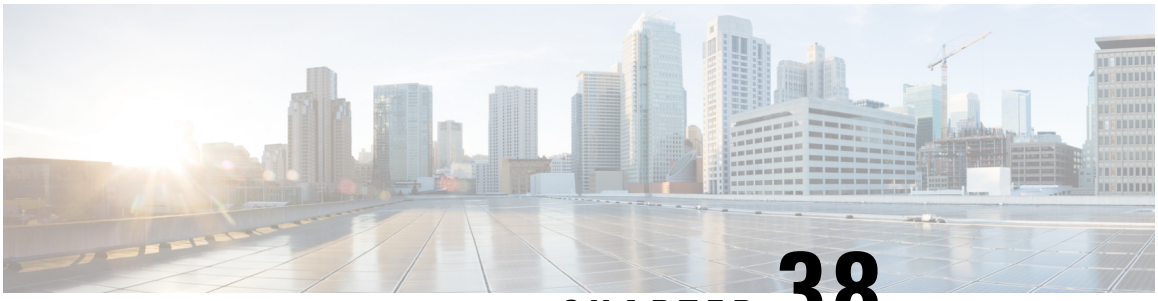
Table 48: Malicious Call Identification Restrictions

Feature	Restriction
Malicious Call Identification Terminating (MCID-T) function	Cisco Unified Communications Manager supports only the malicious call identification originating function (MCID-O). Cisco Unified Communications Manager does not support the malicious call identification terminating function (MCID-T). If Cisco Unified Communications Manager receives a notification from the network of a malicious call identification, Cisco Unified Communications Manager ignores the notification.

Feature	Restriction
Intercluster trunks	MCID does not work across intercluster trunks because Cisco Unified Communications Manager does not support the MCID-T function.
Cisco MGCP FXS gateways	Cisco MGCP FXS gateways do not support MCID. No mechanism exists for accepting the hookflash and collecting the feature code in MGCP.
QSIG trunks	MCID does not work over QSIG trunks because MCID is not a QSIG standard.
Cisco VG248 Analog Phone Gateway	Cisco VG248 Analog Phone Gateway does not support MCID.
SIP trunks	MCID does not support SIP trunks.
Immediate Divert	System does not support using MCID and Immediate Divert features together.

Malicious Call ID Troubleshooting

To track and troubleshoot Malicious Call ID, you can use Cisco Unified Communications Manager SDL traces and alarms. For information about setting traps and traces for MCID, see the *Cisco Unified Serviceability Administration Guide*. For information about how to generate reports for MCID, see the *Cisco Unified CDR Analysis and Reporting Administration Guide*.



CHAPTER 38

Call Transfer

- Call Transfer Overview, on page 495
- Call Transfer Configuration Task Flow, on page 496
- Call Transfer Interactions, on page 506
- Call Transfer Restrictions, on page 507

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

	Command or Action	Purpose
Step 1	Configure Consult and Blind Transfer, on page 496	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.
Step 2	Configure Transfer On-Hook, on page 501	(Optional) Transfer On-Hook is an option to complete call transfers. Press Trnsfer, 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.
Step 3	Configure Direct Transfer, on page 501	(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.

Configure Consult and Blind Transfer

Complete one of the task flows depending on whether your phone supports softkey or buttons.

Procedure

	Command or Action	Purpose
Step 1	Configure a Softkey Template for Transfer, on page 496	
Step 2	Configure Transfer Button, on page 500	

Configure a Softkey Template for Transfer

Trnsfer softkey is used for consult and blind transfer of a call. The trnsfer sofkey has the following call states:

- connected
- on hold

Use this procedure to make the Trnsfer softkey available:

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one the following procedures:

- Associate Transfer Softkey Template with a Common Device Configuration, on page 497
- Associate Transfer Softkey Template with a Phone, on page 499

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 499.

Before you begin

Configure a Softkey Template for Transfer, on page 496

Procedure

	Command or Action	Purpose
Step 1	Add Transfer Softkey Template to the Common Device Configuration, on page 498	Perform this step to add Trnsfer softkey template to the Common Device Configuration.
Step 2	Associate a Common Device Configuration with a Phone, on page 499	Perform this step to link the Trnsfer softkey Common Device Configuration to a phone.

What to do next

Configure Transfer Button, on page 500

Add Transfer Softkey Template to the Common Device Configuration

Before you begin

Configure a Softkey Template for Transfer, on page 496

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Before you begin

Add Transfer Softkey Template to the Common Device Configuration, on page 498

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate Transfer Softkey Template with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Before you begin

Configure a Softkey Template for Transfer, on page 496

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Configure Transfer Button

The procedures in this section describe how to configure the Transfer button.

Procedure

	Command or Action	Purpose
Step 1	Configure a Phone Button Template for Transfer, on page 500	Perform this step to assign Transfer button features to line or speed dial keys.
Step 2	Associate Transfer Button Template with a Phone, on page 500	Perform this step to configure the Transfer button for a phone.

Configure a Phone Button Template for Transfer

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

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate Transfer Button Template with a Phone

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure Transfer On-Hook

Before you begin

Configure Consult and Blind Transfer, on page 496

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 you want to configure the parameter.

Step 3 From the **Service** drop-down list, choose the **Cisco CallManager (Active)** service.

Step 4 In the **Clusterwide Parameters (Device - Phone)**, choose **True** for the **Transfer On-Hook Enabled** service parameter.

Step 5 Click **Save**.

Configure Direct Transfer

Complete one of the task flows depending on whether your phone supports softkey or buttons.

Procedure

	Command or Action	Purpose
Step 1	Configure a Softkey Template for Direct Transfer, on page 501	Perform this step to add Direct Transfer softkey to template and configure the softkey using the Common Device Configuration or phone.
Step 2	Configure Direct Transfer Button, on page 504	Perform this step to add and configure the Direct Transfer button to a phone.

Configure a Softkey Template for Direct Transfer

Direct Transfer softkey has the following call states:

- Connected

- On hold

Use this procedure to make the Direct Transfer softkey available:

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one the following procedures:

- Associate Direct Transfer Softkey Template with a Common Device Configuration, on page 503
- Associate Direct Transfer Softkey Template with a Phone, on page 504

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 504

Before you begin

Configure a Softkey Template for Direct Transfer, on page 501

Procedure

	Command or Action	Purpose
Step 1	Add Direct Transfer Softkey Template to the Common Device Configuration, on page 503	Perform this step to add Direct Transfer softkey template to the Common Device Configuration.
Step 2	Associate a Common Device Configuration with a Phone, on page 504	Perform this step to add Direct Transfer softkey template to the Common Device Configuration.

Add Direct Transfer Softkey Template to the Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

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

- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate a Common Device Configuration with a Phone

Before you begin

Add Direct Transfer Softkey Template to the Common Device Configuration, on page 503

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate Direct Transfer Softkey Template with a Phone

Optional. Use this procedure as an alternative to associating the softkey template with the Common Device Configuration. This procedure also works in conjunction with the Common Device Configuration. You can use it when you need to assign a softkey template that overrides the assignment in the Common Device Configuration or any other default softkey assignment.

Before you begin

Configure a Softkey Template for Direct Transfer, on page 501

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Configure Direct Transfer Button

The procedures in this section describe how to configure the Direct Transfer button.

Procedure

	Command or Action	Purpose
Step 1	Configure Phone Button Template for Direct Transfer, on page 505	Perform this step to assign Direct Transfer button features to line or speed dial keys.
Step 2	Associate Direct Transfer Button Template with a Phone, on page 505	Perform this step to configure the Direct Transfer button for a phone.

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**.

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate Direct Transfer Button Template with a Phone

Before you begin

Configure Phone Button Template for Direct Transfer, on page 505

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

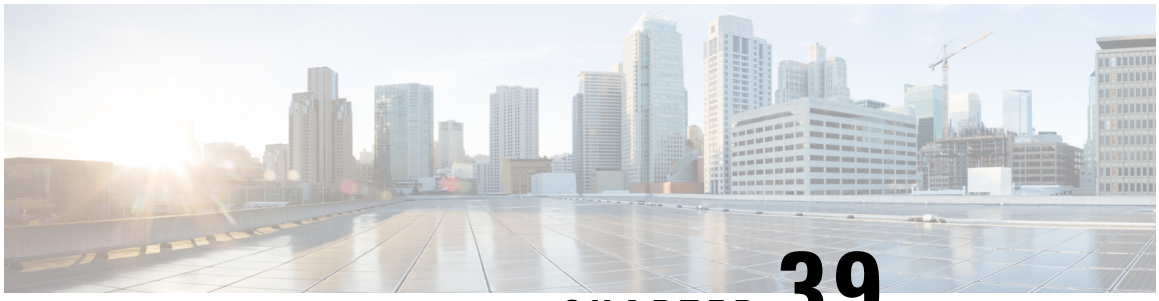
Call Transfer Interactions

Feature	Interaction
Logical Partitioning	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: • When a phone user uses Transfer softkey to transfer the call, the second press of the softkey invokes and processes the Call Transfer feature. • When other transfer mechanisms, such as Direct Transfer, On-Hook Transfer, Hook Flash Transfer, and CTI-application-initiated Transfer results in invoking the Call Transfer feature. • When the transferred and the transferred destination specifies a PSTN participant. • When Cisco Unified Communications Manager uses the geolocation identifier information that associates with the transferred and transferred destination device to perform logical partitioning policy checking. • Before splitting of the primary and secondary calls, and before joining. Logical partitioning handles a denied call as follows: • Sends External Transfer Restricted message to the VoIP phone. • Normal Transfer—For a phone that is running SCCP, the primary call remains on hold, and the consultation call remains active. For a phone that is running SIP, both primary and consultation calls remain on hold and must be resumed manually after the failure. • On-Hook, Hook-Flash and Analog-Phone-Initiated Transfer—Both the primary and secondary calls are cleared by using the cause code=63 “Service or option not available” with a reorder tone from Cisco Unified Communications Manager. • The Number of Transfer Failures perfmon counter is incremented.

Feature	Interaction
Multilevel Precedence and Preemption (MLPP)	When a switch initiates a call transfer between two segments that have the same precedence level, the segments maintain the precedence level upon transfer. When a call transfer is made between call segments that are at different precedence levels, the switch that initiates the transfer marks the connection at the segment that has the higher precedence level. Cisco Unified Communications Manager supports this requirement by upgrading the precedence level of a call leg that is involved in a Call Transfer operation. For example, party A calls party B with Priority precedence level. Party B then initiates a transfer to party C and dials the Flash precedence digits when dialing. When the transfer is complete, the precedence level of party A gets upgraded from Priority to Flash. The Call Transfer feature is enabled automatically when MLPP is enabled, and the phones support the Transfer softkey. **Note** The precedence level upgrade does not work over a trunk device such as an intercluster trunk (ICT) or a PRI trunk.

Call Transfer Restrictions

Feature	Restriction
Logical Partitioning	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.
External Call Transfer Restrictions	To restrict transfer for external call scenarios, see the “External Call Transfer Restrictions” chapter.
Hunt Pilot	If a call transfer to a hunt pilot is initiated when an announcement is in progress, the call is redirected only after the announcement is complete.



CHAPTER 39

External Call Transfer Restrictions

- External Call Transfer Restrictions Overview, on page 509
- Configure External Call Transfer Restrictions Task Flow, on page 510
- External Call Transfer Restrictions Interactions, on page 514
- External Call Transfer Restrictions Restrictions, on page 514

External Call Transfer Restrictions Overview

External Call Transfer Restrictions is a feature that you can use to configure gateways, trunks, and route patterns as OnNet (internal) or OffNet (external) devices at the system level. By setting the devices as OffNet, you can restrict the transferring of an external call to an external device and thus help prevent toll fraud.

If you try to transfer a call on an OffNet gateway or trunk when the service parameter Block OffNet to OffNet Transfer is set to True, a message displays on the user phone to indicate that the call cannot be transferred.

This chapter uses the following terms:

Term	Description
OnNet Device	A device that is configured as OnNet and considered to be internal to the network.
OffNet Device	A device that is considered as OffNet and, when routed, is considered to be external to the network.
Network Location	The location of the device, which is considered as OnNet or OffNet, with respect to the network.
Originating End	The device that gets transferred. The system considers this device as OnNet or OffNet.
Terminating End	The device that receives the transferred call. The system considers this device as OnNet or OffNet.
Incoming Call	A call for which only gateways and trunks call classification settings get used to classify it as OnNet or OffNet. Route Pattern call classification settings do not apply.

Term	Description
Outgoing Call	A call for which the call classification setting of the trunk, gateway, and route pattern gets considered. The Allow Device Override setting on the route pattern determines whether the trunk or gateway call classification setting gets used instead of the route pattern call classification setting.

Configure External Call Transfer Restrictions Task Flow

Procedure

	Command or Action	Purpose
Step 1	Configure the Service Parameter for Call Transfer Restrictions, on page 510	Block external calls from being transferred to another external device or number.
Step 2	To configure incoming calls perform the following procedures: • Configure the Clusterwide Service Parameter, on page 511 • Configure Gateways for Call Transfer Restrictions, on page 512 • Configure Trunks for Call Transfer Restrictions, on page 512	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 513	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 list.

Step 3 Choose **Cisco CallManager (Active)** from the Service drop-down list.

Step 4 Choose **True** from the Block OffNet to OffNet Transfer drop-down list. The default value specifies False.

Step 5 Click **Save**.

Configure Incoming Calls Task Flow

Procedure

	Command or Action	Purpose
Step 1	(Optional) Configure the Clusterwide Service Parameter, on page 511	Configure all gateways or trunks in the Cisco Unified Communications Manager cluster to be OffNet (external) or OnNet (internal).
Step 2	Configure Gateways for Call Transfer Restrictions, on page 512	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 512	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 steps:

Before you begin

Configure the Service Parameter for Call Transfer Restrictions, on page 510

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 list.

Step 3 Choose **Cisco CallManager (Active)** from the Service drop-down list.

Step 4 Choose either OffNet or OnNet (the default specifies OffNet) from the Call Classification drop-down list.

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 511

Procedure

Step 1 From Cisco Unified CM 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 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**.

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 512

Procedure

Step 1 From Cisco Unified CM 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 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 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.**

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 512

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Route/Hunt** > **Route Pattern** and click **Find** to list all route patterns.

Step 2 Choose the route pattern you want to configure, or click **Add New**.

Step 3 In the **Route Pattern Configuration** window, use the following fields to configure transfer capabilities with route pattern configuration:

a) **Call Classification**—Use this drop-down list to classify the call that uses this route Pattern as OffNet or OnNet.

b) **Provide Outside Dial Tone**—If Call Classification is set to OffNet, this check box gets checked.

c) **Allow Device Override**—When this check box is checked, the system uses the Call Classification setting of the trunk or gateway that is associated with the route pattern instead of the Call Classification setting on the Route Pattern Configuration window.

Step 4 Click **Save**.

External Call Transfer Restrictions Interactions

Feature	Interaction
Drop Conference	The Drop Conference feature determines whether an existing ad hoc conference should be dropped by checking whether the conference parties are configured as OffNet or OnNet. You use the service parameter Drop Ad Hoc Conference and choose the option When No OnNet Parties Remain in the Conference to configure the feature. You determine OnNet status for each party by checking the device or route pattern that the party is using. For more information, see topics related to Ad Hoc Conference linking in the "Ad Hoc Conferencing" chapter.
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*.

External Call Transfer Restrictions Restrictions

Restriction	Description
FXS Gateways	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.

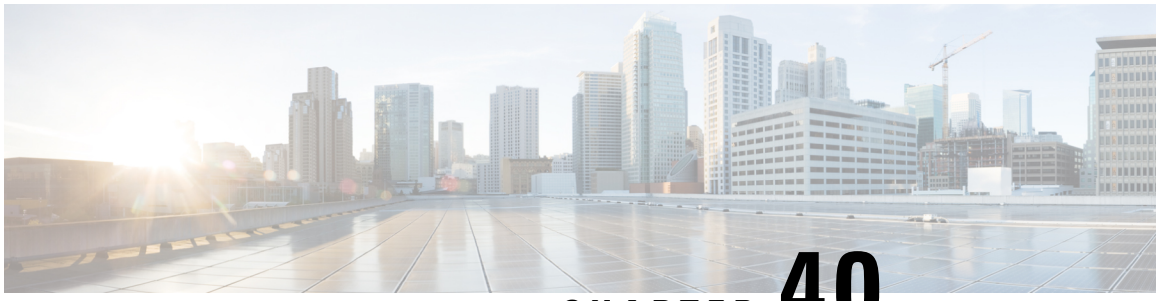
Restriction	Description
Cisco VG248 Gateway	The system does not support the Cisco VG248 Gateway which does not have a Call Classification field.
FXS Ports	Cisco Unified Communications Manager considers all Cisco Unified IP Phones and FXS ports as OnNet (internal) that cannot be configured as OffNet (external).



PART XI

Presence and Privacy Features

- Barge , on page 519
- BLF Presence , on page 531
- Call Display Restrictions , on page 547
- Do Not Disturb , on page 561
- Privacy , on page 575
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CHAPTER 40

Barge

- Barge Overview, on page 519
- Barge Configuration Task Flow, on page 521
- Barge Interactions, on page 528
- Barge Restrictions, on page 529
- Barge Troubleshooting, on page 529

Barge Overview

Barge allows a user to be added to a remotely active call that is on a shared line. Remotely active calls for a line are the active (connected) calls that are made to or from another device that shares a directory number with the line.

If you configure party entrance tone, a tone plays on the phone when a basic call changes to a barged call or 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 Unified Communications Manager failure occurs, the Barge call is preserved.

Note To display the softkey option for both Barge and cBarge, disable the **Privacy** option in Unified Communications Manager user interface for those devices that have shared line appearances.

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 5.

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.

Feature	Barge with Built-In Conference	Barge with Shared Conference
The standard softkey template includes the Barge/cBarge softkey. **Note** If the single button Barge/cBarge feature is enabled, the softkey is not used.	Yes	No
A media break occurs during barge setup.	No	Yes
If configured, a user receives a barge setup tone.	Yes	Yes

Feature	Barge with Built-In Conference	Barge with Shared Conference
Text displays at the barge initiator phone.	To barge XXX	To Conference
Text displays at the target phone.	To/From Other	To Conference
Text displays at the other phones.	To/From Target	To Conference
Bridge supports a second barge setup to an already barged call.	No	Yes
Initiator releases the call.	No media interruption occurs for the two original parties.	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.
Target releases the call.	Media break occurs to reconnect initiator with the other party as a point-to-point call.	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.
Other party releases the call.	All three parties get released.	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.
Target puts call on hold and performs Direct Transfer, Join, or Call Park.	Initiator gets released.	Initiator and the other party remain connected.

Barge Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Configure Softkey Template for Built-In Conferencing, on page 522	Add the Barge softkey to a softkey template. Follow this procedure when you are configuring barge for built-in conference bridges.
Step 2	Configure Softkey Template for Shared Conferencing, on page 523	Add the cBarge softkey to a softkey template. Follow this procedure when you are configuring barge for shared conference bridges.
Step 3	To Associate a Softkey Template with Common Device Configuration, on page 524, complete the following subtasks:	Optional. To make the softkey template available to phones, you must complete either this step or the following step. Follow this step

	Command or Action	Purpose
	• Add a Softkey Template to Common Device Configuration, on page 525 • Associate Common Device Configuration with Phone, on page 526	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 Softkey Template with Phone, on page 524	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 526	Configure barge for built-in conference bridges.
Step 6	Configure Barge for Shared Conferencing, on page 527	Configure barge for shared conference bridges.
Step 7	Associate User with Device, on page 60	Associate users with devices.

Configure Softkey Template for Built-In Conferencing

Configure a softkey template for Barge and assign the Barge softkey to that template. You can configure the Barge softkey in the **Remote In Use** call state.

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one of the following procedures:

- Add a Softkey Template to Common Device Configuration, on page 525
- Associate Common Device Configuration with Phone, on page 526

Configure Softkey Template for Shared Conferencing

Configure a softkey template for shared conferencing and assign the cBarge softkey to that template. You can configure the cBarge softkey in the **Remote In Use** call state.

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

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 524.

Procedure

Step 1 Add a Softkey Template to Common Device Configuration, on page 372

Step 2 Associate a Common Device Configuration with a Phone, on page 372

Add a Softkey Template to Common Device Configuration

Before you begin

Perform one or both of the following as needed:

- Configure Softkey Template for Built-In Conferencing, on page 522
- Configure Softkey Template for Shared Conferencing, on page 523

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.
b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.

a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate Common Device Configuration with Phone

Before you begin

Perform one or both of the following as needed:

- Configure Softkey Template for Built-In Conferencing, on page 522
- Configure Softkey Template for Shared Conferencing, on page 523

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

What to do next

Perform one or both of the following:

- Configure Barge for Built-In Conferencing, on page 526
- Configure Barge for Shared Conferencing, on page 527

Configure Barge for Built-In Conferencing

Most Cisco Unified IP Phones include the built-in conference bridge capability; that is, these Cisco IP Phones have an internal DSP that acts as a small conference bridge to support the barge feature. It can support only a maximum of three parties that include the phone itself. Starting from firmware version 11.x, Cisco IP Phone 8800 Series have the capability to daisy chain the built-in bridge (BIB) feature.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Service Parameters** and set the **Built In Bridge Enable** clusterwide service parameter to **On**.

Note
If this parameter is set to **Off**, configure barge for each phone by setting the **Built in Bridge** field in the **Phone Configuration** window.

Step 2 Set the **Party Entrance Tone** clusterwide service parameter to **True** (required for tones) or configure the **Party Entrance Tone** field in the **Directory Number Configuration** window.

Step 3 Set the **Single Button Barge/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

Before you begin

Perform one or both of the following:

- Configure Barge for Built-In Conferencing, on page 526
- Configure Barge for Shared Conferencing, on page 527

Procedure

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

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

Step 3 Select the user from the list.
The **End User Configuration** window appears.

Step 4 Locate the **Device Information** section.

Step 5 Click **Device Association**.
The **User Device Association** window appears.

Step 6 Find and select the CTI remote device.

Step 7 To complete the association, click **Save Selected/Changes**.

Step 8 From **Related Links** drop-down list, choose **Back to User**, and then click **Go**.
The **End User Configuration** window appears, and the associated device that you chose appears in the **Controlled Devices** pane.

Barge Interactions

Feature	Interaction
cBarge	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. **Note** You can enable Single-Button Barge or Single-Button cBarge for a device, but not both.
Call Park	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).
Join	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).
Private Line Automatic Ringdown (PLAR)	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.

Barge Restrictions

Restriction	Description
Additional callers	The Barge initiator cannot conference in additional callers.
Computer Telephony Interface (CTI)	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.
G.711 codec	The original call requires G.711 codec. If G.711 is not available, use cBarge instead.
Cisco Unified IP Phones	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.
Encryption	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.
Maximum number of calls	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`.

Barge Troubleshooting

No Conference Bridge Available

When the Barge softkey is pressed, the message `No Conference Bridge Available` is displayed on the IP phone.

The **Built In Bridge** field in the **Phone Configuration** window for the target phone is not set properly.

To resolve the problem, perform the following steps:

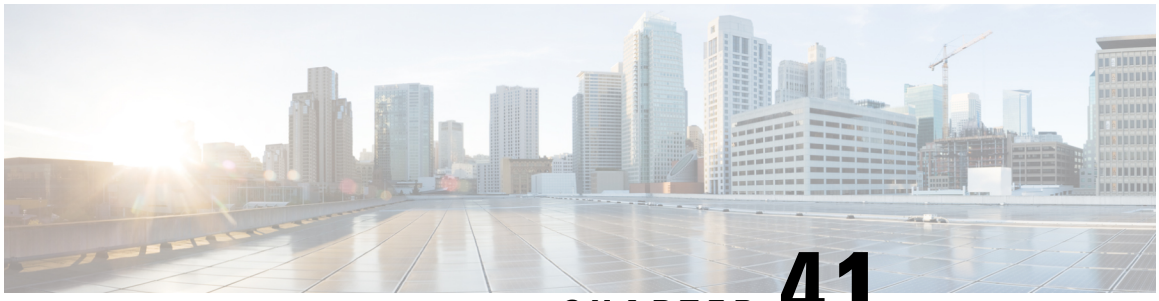
1. From Cisco Unified CM Administration, choose **Device** > **Phone** and click **Find the phone** to find the phone configuration of the phone that is having the problem.
2. Set the **Built In Bridge** field to **On**.
3. Click **Update**.
4. Reset the phone.

Error: Past Limit

The phone displays the message, `Error: Past Limit`.

The number of shared-line users in the conference is equal to or greater than the configuration for the **Maximum Number of Calls** field for the device from which you are attempting to barge.

- Go to **Service Parameter Configuration** window and locate the **Clusterwide Parameters (Feature - Conference)** section. Increase the value of **Maximum Ad Hoc Conference** parameter as required.
- Check the **Maximum Number of Calls** value for the shared lines on the device from which you are attempting to barge and increase the value as required.



CHAPTER 41

BLF Presence

- BLF Presence Overview, on page 531
- BLF Presence Prerequisites, on page 531
- BLF Presence Configuration Task Flow, on page 532
- BLF Presence Interactions, on page 543
- BLF Presence Restrictions, on page 543

BLF Presence Overview

The Busy Lamp Field (BLF) presence feature allows a user who is a watcher to monitor the real-time status of another user at a directory number or Session Initiation Protocol (SIP) uniform resource identifier (URI) from the device of the watcher.

A watcher can monitor the status of the user or BLF presence entity (also called presentity) by using the following options:

- BLF and SpeedDial buttons
- Missed call, placed call, or received call lists in the directories window
- Shared directories, such as the corporate directory

Call lists and directories display the BLF status for existing entries. When you configure BLF and SpeedDial buttons, the BLF presence entity appears as a speed dial on the device of the watcher.

To view the status of a BLF presence entity, watchers send BLF presence requests to Cisco Unified Communications Manager. After administrators configure BLF presence features, real-time status icons appear on the watcher device to indicate whether the BLF presence entity is on the phone, is not on the phone, the status is unknown, and so on.

Extension mobility users can use BLF presence features on phones with extension mobility support.

BLF presence group authorization ensures that only authorized watchers can access the BLF presence status for a destination. Because the administrator ensures that the watcher is authorized to monitor the destination when a BLF or Speed Dial is configured, BLF presence group authorization does not apply to BLF or Speed Dials.

BLF Presence Prerequisites

- Configure the phones that you want to use with the BLF presence feature.

- Configure the SIP trunks that you want to use with the BLF presence feature.

BLF Presence Configuration Task Flow

Before you begin

- Review BLF Presence Prerequisites, on page 531

Procedure

	Command or Action	Purpose
Step 1	Configure and synchronize cluster-wide enterprise parameters for Busy Lamp Field (BLF). See Configure/Synchronize Cluster-Wide Enterprise Parameters for BLF, on page 533.	Configure BLF options that apply to all devices and services in the same cluster. You can synchronize enterprise-parameter configuration changes with the configured devices in the least-intrusive manner. For example, a reset or restart may not be required on some affected devices.
Step 2	Configure cluster-wide service parameters for BLF. See Configure Cluster-Wide Service Parameters for BLF, on page 534.	Configure presence service parameters to configure different services on selected servers in Cisco Unified Communications Manager Administration.
Step 3	Configure BLF presence groups. See Configure BLF Presence Groups, on page 534.	Configure BLF presence groups to control the destinations that watchers can monitor.
Step 4	To associate BLF presence group with devices and users, perform the following subtasks: • Associate BLF presence groups with phones. See Associate BLF Presence Groups with Phone, on page 536. • Associate BLF presence groups with SIP trunks. See Associate BLF Presence Groups with SIP Trunk, on page 537. • Associate BLF presence groups with an end user. See Associate BLF Presence Groups with End User, on page 538. • Associate BLF presence groups with an application user. See Associate BLF Presence Groups with Application User, on page 539.	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 540.	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 a Calling Search Space for Presence Requests, on page 540.	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 541.	Configure a phone button template for BLF and SpeedDial buttons 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 542.	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 542.	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 From Cisco Unified CM Administration, choose **System** > **Enterprise Parameters**.

Step 2 Configure the fields in the **Enterprise Parameters Configuration** window. For more information on the fields and their configuration options, see Online Help.

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 533

Procedure

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

Step 2 From the **Server** drop-down list, choose the server where you want to configure the parameter.

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

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, 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 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 534

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **BLF Presence Group**.

Step 2 Configure the fields in the **BLF Presence Group Configuration** window. See BLF Presence Group Fields for BLF, on page 535 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 536
- Associate BLF Presence Groups with SIP Trunk, on page 537
- Associate BLF Presence Groups with End User, on page 538
- Associate BLF Presence Groups with Application User, on page 539

BLF Presence Group Fields for BLF

Presence authorization works with BLF presence groups. The following table describes the BLF presence group configuration fields.

Field	Description
Name	Enter the name of the BLF presence group that you want to configure. For example, Executive_Group.
Description	Enter a description for the BLF presence group that you are configuring.
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 colocated

Before you begin

Configure BLF Presence Groups, on page 534

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 537
- Associate BLF Presence Groups with End User, on page 538
- Associate BLF Presence Groups with Application User, on page 539

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, 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 534

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Trunk**, and click **Add New**.

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 536
- Associate BLF Presence Groups with End User, on page 538
- Associate BLF Presence Groups with Application User, on page 539

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 534

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 536
- Associate BLF Presence Groups with SIP Trunk, on page 537
- Associate BLF Presence Groups with Application User, on page 539

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 534

Procedure

Step 1 In the Cisco Unified CM Administration, choose **User Management** > **Application User**, and click **Add New**.
The **Application User Configuration** window appears.

Step 2 Configure the fields in the **Application User Configuration** window. See the online help for information about the fields and their configuration options.

Step 3 Click **Save**.

What to do next

Associate BLF presence group with devices and users by performing the following subtasks:

- Associate BLF Presence Groups with Phone, on page 536
- Associate BLF Presence Groups with SIP Trunk, on page 537
- Associate BLF Presence Groups with End User, on page 538

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 536
- Associate BLF Presence Groups with SIP Trunk, on page 537
- Associate BLF Presence Groups with End User, on page 538
- Associate BLF Presence Groups with Application User, on page 539

Procedure

Step 1 From 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.

Configure a Calling Search Space for Presence Requests

The SUBSCRIBE Calling Search space option allows you to apply a calling search space separate from the call-processing Calling Search Space for BLF presence requests. Select a different calling search space for presence requests, else the SUBSCRIBE Calling Search Space selects the **None** default option. The SUBSCRIBE Calling Search Space that is associated with an end user is used for extension mobility calls.

You apply the SUBSCRIBE Calling Search Space to the SIP trunk, phone, or end user. The SUBSCRIBE Calling Search Space that is associated with an end user is used for extension mobility calls.

Before you begin

Accept BLF Presence Requests from External Trunks and Applications, on page 540

Procedure

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

Step 2 In the **Calling Search Space configuration** window, choose the calling search space from the **SUBSCRIBE Calling Search Space** drop-down list.

Step 3 Click **Add New**.

Step 4 In the **Name** field, enter a name.

Step 5 (Optional) In the **Description** field, enter a description to identify the calling search space.

Step 6 From the **Available Partitions** list, select one or multiple partitions, and click the arrow keys.
The selected partitions appear in the **Selected Partitions** list.

Step 7 (Optional) To add or remove a partition from the **Selected Partitions** list, click the arrow keys next to the list box.

Step 8 Click **Save**.

All calling search spaces that you configure in Cisco Unified Communications Manager Administration appear in the **SUBSCRIBE Calling Search Space** drop-down list in the **Trunk Configuration** or **Phone Configuration** window.

Configure a Phone Button Template for BLF and SpeedDial Buttons

You can configure BLF and SpeedDial buttons for a phone or user device profile. After you apply the template to the phone or device profile (and save the phone or device profile configuration), the Add a new BLF SD link appears in the **Association Information** pane in Cisco Unified Communications Administration.

Note If the template does not support BLF and SpeedDials, the Add a new BLF SD link appears in the **Unassigned Associated Items** pane.

When an administrator decides to add or change a BLF and SpeedDial button for a SIP URI, the administrator ensures that the watcher is authorized to monitor that destination. If the system uses a SIP trunk to reach a SIP URI BLF target, the BLF presence group associated with the SIP trunk applies.

Note You do not need to configure BLF presence groups or the Default Inter-Presence Group Subscription parameter for BLF and SpeedDials.

Before you begin

Configure a Calling Search Space for Presence Requests, on page 540

Procedure

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

Step 2 Click the **Add New** button.
The **Phone Button Template Configuration** window appears.

Step 3 In the **Button Template Name** field, enter a name for the template.

Step 4 From the **Phone Button Template** drop-down list, select a template of phone button.

Step 5 Click **Copy** to create a new button template based on the layout of the selected button template.

Step 6 Click **Save**.

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 Unified Communications Manager.

Before you begin

Configure a Phone Button Template for BLF and SpeedDial Buttons, on page 541

Procedure

Step 1 From 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**.

Configure User Device Profile

See the “BLF Presence with Extension Mobility” section of BLF Presence Interactions, on page 543 for details.

Before you begin

Associate Button Template with a Device, on page 542

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

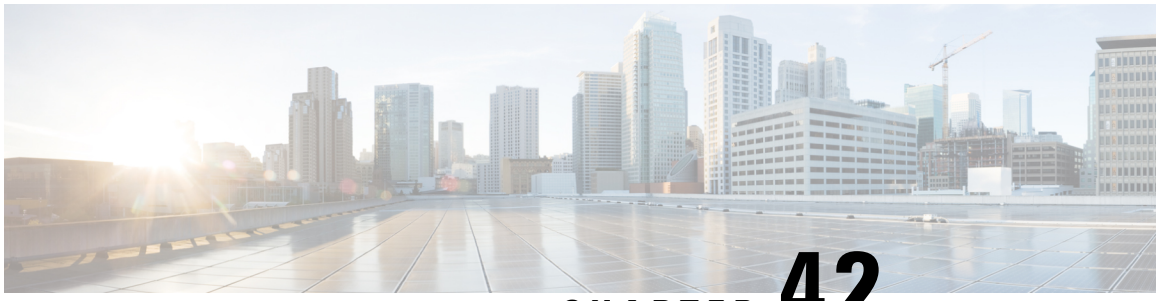
Feature	Interaction
Presence BLF with DNs on H.323 phones when the H.323 phone device serves as presence entity	When the H.323 phone is in the RING IN state, the BLF status gets reported as Busy. For the presence entities of phones that are running either SCCP or SIP and that are in the RING IN state, the BLF status gets reported as Idle.
Presence BLF with DNs on H.323 phones when the H.323 phone device serves as presence entity	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.
BLF Presence with Extension Mobility	When you configure BLF and SpeedDial buttons in a user device profile in Cisco Unified Communications Manager Administration, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF and SpeedDial buttons after you log in to the device. When the extension mobility user logs out, a phone that supports Cisco Extension Mobility displays BLF presence status on the BLF and SpeedDial buttons for the logout profile that is configured.

BLF Presence Restrictions

Restriction	Description
SIP Presence	Cisco Unified Communications Manager Assistant does not support SIP presence.

Restriction	Description
BLF Presence Requests	Cisco Unified Communications Manager Administration rejects BLF presence requests to a directory number that is associated with a hunt pilot.
BLF on Call List Feature	The BLF on call list feature is not supported on the Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7960.
BLF and SpeedDials	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. **Note** BLF presence group authorization does not apply to any directory number or SIP URI that is configured as a BLF and Speed Dial that appears in a call list for phones that are running SIP. If there is an overlapping DN, where there is the same extension in different partitions, the presence notifications are selected based on the order of the partitions configured within the SUBSCRIBE CSS assigned to the device. For example, two BLF speed dials are configured on a phone. • Extension 1234 in the "internal" partition • Extension 1234 in the "external" partition Whichever partition is listed first within the SUBSCRIBE CSS is the one that will provide BLF presence to the subscribed devices.
BLF Presence Authorization	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.
Cisco Unified IP Phone	When a user monitors a directory number that is configured for Cisco Unified IP Phones 7960 and 7940 that are running SIP, the system displays a status icon for ‘not on the phone’ on the watcher device when the presence entity is off-hook (but not in a call connected state). These phones do not detect an off-hook status. For all other phone types, the system displays the status icon for ‘on the phone’ on the watcher device for an off-hook condition at the presence entity.
SIP Trunks	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.
BLF Presence-supported Phones that are running SIP	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.

Restriction	Description
Phones that are running SIP	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.



CHAPTER 42

Call Display Restrictions

- Call Display Restrictions Overview, on page 547
- Call Display Restrictions Configuration Task Flow, on page 547
- Call Display Restrictions Interactions, on page 557
- Call Display Restrictions Feature Restrictions, on page 559

Call Display Restrictions Overview

Cisco Unified Communications Manager provides flexible configuration options that allow and also restrict the display of the number and name information for both calling and connected users. You can restrict connected numbers and names independently of each other.

You can configure connected number and name restrictions on the SIP trunk level or on a call-by-call basis. The SIP trunk level configuration overrides a call-by-call configuration.

For example, in a hotel environment, you may want to see the display information for calls that are made between a guest room and the front desk. However, for calls between guest rooms, you can restrict the call information to display on either phone.

Call Display Restrictions Configuration Task Flow

Before you begin

- Review Call Display Restrictions Interactions, on page 557

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify endpoints that support the Call Display Restrictions feature.
Step 2	Configure Partitions for Call Display Restrictions, on page 548	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

	Command or Action	Purpose
		configure a partition for dialing between rooms, and a partition for dialing the public switched telephone network (PSTN).
Step 3	Configure Calling Search Spaces for Call Display Restrictions, on page 549.	Configure calling search spaces to identify the partitions that calling devices can search when they attempt to complete a call. Create calling search spaces for rooms, the front desk, other hotel extensions, the PSTN, and the room park range (for call park).
Step 4	Configure the Service Parameter for Connected Number Display Restriction, on page 550.	Configure the service parameter to display the connected line ID as dialed digits only.
Step 5	Configure Translation Patterns, on page 551.	Configure translation patterns with different levels of display restrictions.
Step 6	Configure Phones for Call Display Restrictions, on page 552	Associate endpoints with the partitions and the calling search spaces that you want to use for call display restrictions.
Step 7	Configure the PSTN Gateway for Call Display Restrictions, on page 555	Associate the PSTN gateway with the partitions and the calling search spaces that you want to use for call display restrictions.
Step 8	Optional. Configure Call Display Restrictions on SIP Trunks, on page 555	Use this procedure to configure connected number and name restrictions on the SIP trunk level. The SIP trunk level configuration overrides call-by-call configuration.

Configure Partitions for Call Display Restrictions

Configure partitions to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Partitions facilitate call routing by dividing the route plan into logical subsets that are based on organization, location, and call type. You can configure multiple partition

Procedure

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

Step 2 Click **Add New** to create a new partition.

Step 3 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 4 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 5 To create multiple partitions, use one line for each partition entry.

Step 6 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 7 Select one of the following radio buttons to configure the **Time Zone**:

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

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 49: Partition Name Guidelines

Partition Name Length	Maximum Number of Partitions
2 characters	340
3 characters	256
4 characters	204
5 characters	172
...	...
10 characters	92
15 characters	64

Configure Calling Search Spaces for Call Display Restrictions

Configure calling search spaces to identify the partitions that calling devices can search when they attempt to complete a call. Create calling search spaces for rooms, the front desk, other hotel extensions, the PSTN, and the room park range (for call park).

Before you begin

Configure Partitions for Call Display Restrictions, on page 548

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

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

Step 4 In the **Description** field, enter a description.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

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

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

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

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

Step 8 Click **Save**.

Configure the Service Parameter for Connected Number Display Restriction

The connected number display restriction restricts the connected line ID display to dialed digits only. This option addresses customer privacy issues as well as connected number displays that are meaningless to phone users.

Before you begin

Configure Calling Search Spaces for Call Display Restrictions, on page 549

Procedure

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

Step 2 Select the server where the Cisco CallManager service runs, and then select the Cisco CallManager service.

Step 3 Set the **Always Display Original Dialed Number** service parameter to **True** to enable this feature.

The default value is **False**.

Step 4 (Optional) Set the **Name Display for Original Dialed Number When Translated** service parameter.

The default field shows the alerting name of the original dialed number before translation. You can change this parameter to show the alerting name of the dialed number after translation. This parameter is not applicable if the **Always Display Original Number** service parameter is set to **False**.

Step 5 Click **Save**.

Configure Translation Patterns

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 550

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Translation Pattern**.

Step 2 Configure the fields in the **Translation Pattern Configuration** window. See Translation Pattern Fields for Call Display Restrictions, on page 551 for more information about the fields and their configuration options.

Step 3 Click **Save**.

Translation Pattern Fields for Call Display Restrictions

Field	Description
Translation Pattern	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 +.
Description	Enter a description for the translation pattern. The description can include up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), or angle brackets (<>).
Partition	From the drop-down list, choose the partition to associate with this translation pattern.
Calling Search Space	From the drop-down list, choose the calling search space to associate with this translation pattern.

Field	Description
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.
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.

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 551

Procedure

Step 1 From 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 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

Scenario—Hotel Zone Configuration

A hotel environment is segmented into three zones: **Guest Rooms**, **Front Desk**, and **Club**. Each zone is assigned a unique partition and Calling Search Space (CSS). Front Desk phones are configured to ignore presentation restrictions to ensure that staff can always identify the caller.

Configuration:

| Phone | Location | Directory Number | Partition | Calling Search Space | Ignore Presentation Indicators |

|----------|----------|----------|----------|----------|----------|

| A | Room-1 | 221 | `P_Room` | `CSS_FromRoom` | No |

| B | Room-2 | 222 | `P_Room` | `CSS_FromRoom` | No |

| C | Front Desk-1 | 100 | `P_FrontDesk` | `CSS_FromFrontDesk` | **Yes** |

| D | Front Desk-2 | 200 | `P_FrontDesk` | `CSS_FromFrontDesk` | **Yes** |

| E | Club | 300 | `P_Club` | `CSS_FromClub` | No |

Key takeaways:

- Phones within the same zone share a common partition and CSS (e.g., `P_Room` and `CSS_FromRoom`; `P_FrontDesk` and `CSS_FromFrontDesk`).
- Front Desk phones have the **Ignore Presentation Indicators** setting enabled. This ensures that the caller name and number are displayed to reception staff, even when the caller has restricted their information.
- The combination of partitions and CSS on each device governs reachable destinations and the display of caller or connected identity.

Scenario—Clinic Environment Configuration

A clinic environment utilizes three specific phone configurations to manage caller identity visibility:

- **Doctor Phones (Consulting Rooms)**: Configured to maintain privacy; external parties cannot view the doctor’s name or number.
- **Reception Phone**: Configured to override privacy settings; the receptionist always sees the real caller identity to facilitate accurate call transfers.
- **Waiting Room Phone**: Used by patients; internal caller identities are masked to ensure security.

Configuration:

| Phone | Extension | Partition | Calling Search Space (CSS) | Ignore Presentation Indicators |

|---|---|---|---|---|

| Doctor-1 | 201 | `P_Doctor` | `CSS_Doctor` | No |

| Doctor-2 | 202 | `P_Doctor` | `CSS_Doctor` | No |

| Reception | 100 | `P_Reception`| `CSS_Reception` | **Yes** |

| Waiting Room | 300 | `P_Public` | `CSS_Public` | No |

Definition of Terms

Partition: A logical grouping of directory numbers (e.g., Doctor, Reception, Public).

Calling Search Space (CSS): An ordered list of partitions that determines which destinations a phone can reach.

Ignore Presentation Indicators: A device-level setting that allows the phone to display the actual caller name and number, regardless of the caller's privacy status.

Call Behavior Analysis

- **Doctor-to-Doctor Calls**: Both devices reside in P_Doctor with CSS_Doctor. Caller identity information is displayed on both devices as standard.
- **Private Doctor to Reception**: The Reception phone displays the caller's identity ("Doctor-1, 201") because Ignore Presentation Indicators is enabled.

- **Waiting Room to Doctor**: The call connects; however, the Waiting Room phone does not display the doctor's identity, as CSS_Public does not provide visibility into the P_Doctor partition for identity information.
- **Doctor to Waiting Room**: The Waiting Room screen displays a generic label (e.g., "Clinic") to ensure the doctor's identity remains protected from patients.

This configuration uses three partitions, three CSSs, and a single check box to manage complex call display requirements, offering an efficient alternative to traditional Call Display Restrictions.

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 552

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 list.

Gateway Configuration Example

Configure PSTN Gateway E with route pattern P_PSTN and calling search space CSS_FromPSTN

{CSS_FromPSTN}, RoutePattern {P_PSTN}

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 555

Procedure

Step 1 From 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 SIP Trunk Fields for Call Display Restrictions, on page 556 for more information about the fields and their configuration options.

Step 5 Click **Save**.

SIP Trunk Fields for Call Display Restrictions

Table 50: Inbound Calls

Field	Description
Calling Line ID Presentation	From the drop-down list, choose one of the following options: • **Default**—Choose this option if you do not want to change the presentation of the calling line ID. • **Allowed**—Choose this option if you want to display the phone number of the calling party. • **Restricted**—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling party phone number.
Calling Name Presentation	From the drop-down list, choose one of the following options: • **Default**—Choose this option if you do not want to change the presentation of the calling name. • **Allowed**—Choose this option if you want to display the name of the calling party. • **Restricted**—Choose this option if you want Cisco Unified Communications Manager to block the display of the calling name.
Calling Search Space	From the drop-down list, choose the calling search space to associate with this translation pattern.

Table 51: Outbound Calls

Field	Description
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.

Call Display Restrictions Interactions

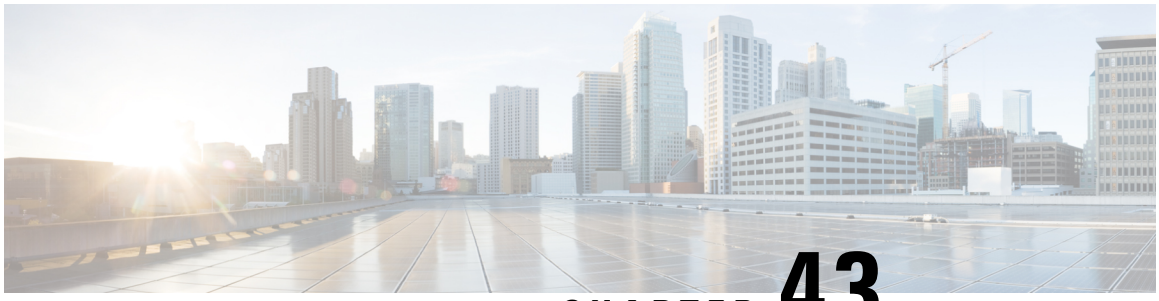
This section describes how the Call Display Restrictions feature interacts with Cisco Unified Communications Manager applications and call processing features.

Feature	Interaction
Call Park	When you use the Call Display Restrictions feature with Call Park, you must configure an associated translation pattern for each individual call park number to preserve the Call Display Restrictions feature. You cannot configure a single translation pattern to cover a range of call park numbers. Consider the following scenario as an example: 1. The system administrator creates a call park range of 77x and places it in a partition called P_ParkRange. (The phones in the guest rooms can see that the P_ParkRange partition is made visible to the phones in the guest rooms by inclusion of it in the calling search space of the phones [CSS_FromRoom]). 2. The administrator configures a separate translation pattern for each call park directory number and configures the display fields to Restricted. (In the current scenario, the administrator creates translations patterns for 770, 771, 772...779.) **Note** For the Call Display Restrictions feature to work correctly, the administrator must configure separate translation patterns and not a single translation pattern for a range of numbers (such as 77x or 77[0-9]). 3. Room-1 calls Room-2. 4. Room-2 answers the call, and Room-1 parks the call. 5. When Room-1 retrieves the call, Room-2 does not see Room-1 call information display. See Call Park and Directed Call Park
Conference List	When you use Call Display Restrictions, you restrict the display information for the list of participants in a conference. See Ad Hoc Conferencing
Conference and Voice Mail	When you use Call Display Restrictions with features, such as conference and voice mail, the call information display on the phones reflects that status. For example, when the conference feature is invoked, the call information display shows **To Conference**. When voice mail is accessed by choosing the Messages button, the call information display shows **To Voicemail**.
Extension Mobility	To use Call Display Restrictions with Extension Mobility, enable the **Ignore Presentation Indicators (internal calls only)** parameter in both the Cisco Unified Communications Manager Administration Phone Configuration window and the Cisco Unified Communications Manager Administration Device Profile Configuration window. When you enable Call Display Restrictions with Extension Mobility, the presentation or restriction of the call information depends on the line profile that is associated with the user who is logged in to the device. The configuration that is entered in the user device profile (associated with the user) overrides the configuration that is entered in the phone configuration (of the phone that is enabled for Extension Mobility).

Feature	Interaction
Call Forwarding	The Connected Number Display restriction applies to all calls that originate in the system. When this value is set to **True**, this field interacts with existing Cisco Unified Communications Manager applications, features, and call processing. This value applies to all calls that terminate inside or outside the system. The Connected Number Display is updated to show the modified number or redirected number when a call is routed to a Call Forward All or Call Forward Busy destination, or gets redirected through a call transfer or CTI application.

Call Display Restrictions Feature Restrictions

Translation Patterns—Duplicate entries are not allowed in translation patterns.



CHAPTER 43

Do Not Disturb

- Do Not Disturb Overview, on page 561
- Do Not Disturb Configuration Task Flow, on page 562
- Do Not Disturb Interactions and Restrictions, on page 570
- Do Not Disturb Troubleshooting, on page 572

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

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	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.
Step 2	Configure Busy Lamp Field Status, on page 563	Configure the Busy Lamp Field status service parameter.
Step 3	Configure Do Not Disturb on a Common Phone Profile, on page 563	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 564.	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 565 • Configure a Do Not Disturb Softkey, on page 567	Add a Do Not Disturb feature button or softkey to your phone.

Configure Busy Lamp Field Status

You can configure how the BLF status depicts Do Not Disturb by setting the **BLF Status Depects DND** service parameter. To set the BLF status, do the following:

Before you begin

Note

- Busy Lamp Field (BLF) presence status for DND works only when all the registered devices for that shared line DN are set to DND.
- If you're using Jabber for iOS or Jabber for Android on the same DN, they are considered registered, even when they are not registered but just configured.

Generate a Phone Feature List, on page 5

Procedure

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

Step 2 Choose the **Cisco CallManager** service for the server that you want to configure.

Step 3 In the Clusterwide Parameters (System - Presence) pane, specify one of the following values for the **BLF Status Depicts DND** service parameter:

- **True**—If Do Not Disturb is activated on the device, the BLF status indicator for the device or line appearance reflects the Do Not Disturb state.
- **False**—If Do Not Disturb is activated on the device, the BLF status indicator for the device or line appearance reflects the actual device state.

What to do next

Perform one of the following procedures:

Configure Do Not Disturb on a Common Phone Profile, on page 563

Apply Do Not Disturb Settings to the Phone, on page 564

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 563

Procedure

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

Step 2 From the **DND Option** drop-down list, 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, 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**.

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 563.

Otherwise, complete Configure Busy Lamp Field Status, on page 563

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, 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, 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, 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 565

Configure a Do Not Disturb Softkey, on page 567

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

	Command or Action	Purpose
Step 1	Configure Phone Button Template for Do Not Disturb, on page 566	Create a phone button template that includes the Do Not Disturb button.
Step 2	Associate a Button Template with a Phone, on page 251	Associate the Do Not Disturb button template to a phone.

Configure Phone Button Template for Do Not Disturb

Follow this procedure to configure a phone button template that includes the Do Not Disturb button.

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate Button Template with Phone

Before you begin

Configure Phone Button Template for Do Not Disturb, on page 566

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure 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

	Command or Action	Purpose
Step 1	Configure Softkey Template for Do Not Disturb, on page 567	Create a softkey template that includes the Do Not Disturb softkey.
Step 2	Perform either of the following procedures: • Associate a Softkey Template with a Common Device Configuration, on page 568 • Associate Softkey Template with a Phone, on page 569	You can associate the softkey to a Common Device Configuration and then associate that configuration to a group of phones, or you can associate the softkey template directly to a phone.

Configure Softkey Template for Do Not Disturb

Perform these steps to configure a softkey template that includes the Do Not Disturb softkey.

Procedure

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

Step 2 Perform the following steps to create a new softkey template; otherwise, proceed to the next step.

a) Click **Add New**.
b) Select a default template and click **Copy**.
c) Enter a new name for the template in the **Softkey Template Name** field.
d) Click **Save**.

Step 3 Perform the following steps to add softkeys to an existing template.

a) Click **Find** and enter the search criteria.
b) Select the required existing template.

Step 4 Check the **Default Softkey Template** check box to designate this softkey template as the default softkey template.

Note
If you designate a softkey template as the default softkey template, you cannot delete it unless you first remove the default designation.

Step 5 Choose **Configure Softkey Layout** from the **Related Links** drop-down list in the upper right corner and click **Go**.

Step 6 From the **Select a Call State to Configure** drop-down list, choose the call state for which you want the softkey to display.

Step 7 From the **Unselected Softkeys** list, choose the softkey to add and click the right arrow to move the softkey to the **Selected Softkeys** list. Use the up and down arrows to change the position of the new softkey.

Step 8 Repeat the previous step to display the softkey in additional call states.

Step 9 Click **Save**.

Step 10 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them. For more information, see *Add a Softkey Template to a Common Device Configuration* and *Associate a Softkey Template with a Phone* sections.

What to do next

Perform one of the following procedures to add the softkey template to a phone.

Associate a Softkey Template with a Common Device Configuration, on page 568

Associate Softkey Template with a Phone, on page 569

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 567

Procedure

	Command or Action	Purpose
Step 1	Add Softkey Template to Common Device Configuration, on page 568	Associate the DND softkey template to a Common Device Configuration.
Step 2	Associate Common Device Configuration with Phone, on page 569	Add the DND softkey to a phone by associating the Common Device Configuration to the phone.

Add Softkey Template to Common Device Configuration

Procedure

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

Step 2 Perform the following steps to create a new Common Device Configuration and associate the softkey template with it; otherwise, proceed to the next step.

a) Click **Add New**.

b) Enter a name for the Common Device Configuration in the **Name** field.
c) Click **Save**.

Step 3 Perform the following steps to add the softkey template to an existing Common Device Configuration.
a) Click **Find** and enter the search criteria.
b) Click an existing Common Device Configuration.

Step 4 In the **Softkey Template** drop-down list, choose the softkey template that contains the softkey that you want to make available.

Step 5 Click **Save**.

Step 6 Perform one of the following tasks:

- If you modified a Common Device Configuration that is already associated with devices, click **Apply Config** to restart the devices.
- If you created a new Common Device Configuration, associate the configuration with devices and then restart them.

Associate Common Device Configuration with Phone

Before you begin

Associate a Softkey Template with a Common Device Configuration, on page 568

Procedure

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

Step 2 Click **Find** and select the phone device to add the softkey template.

Step 3 From the **Common Device Configuration** drop-down list, choose the common device configuration that contains the new softkey template.

Step 4 Click **Save**.

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

Associate 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 567

Procedure

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

Step 2 Click **Find** to select the phone to add the softkey template.

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

Step 4 Click **Save**.

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

Do Not Disturb Interactions and Restrictions

This section provides information about Do Not Disturb interactions and restrictions.

Interactions

The following table describes feature interactions with the Do Not Disturb (DND) feature. Unless otherwise stated, the interactions apply to both the DND Ringer Off and the DND Call Reject option.

Feature	Interaction with Do Not Disturb
Call Forward All	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.
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.

Feature	Interaction with Do Not Disturb
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 ring is supported on both SCCP and SIP.
Call Back	For the originating side, callback overrides DND. When the activating device is on DND mode, the callback notification (both audio and visual) is still presented to the user. For the terminating side, DND overrides callback as follows: • When the terminating side is on DND Ringer Off, the Callback Available screen is sent after the terminating side goes off hook and on hook. • When the terminating side is on DND Call Reject, and is available, a new screen is sent to the activating device as "<DirectoryNumber> has become available but is on DND-R" if the activating device is in same cluster. Callback available notification is sent only after the terminating side disables DND Call Reject.
Pickup Notification	For the DND Ringer Off option, only visual notification gets presented to the device. For the DND Call Reject option, no notification gets presented to the device.
Hunt List	If a device in a Hunt List has DND Ringer Off activated, the call is still presented to the user. However, the DND Incoming Call Alert settings would still apply. If a device in a Hunt List has DND Call Reject activated, any calls to that Hunt List will go to the next member and will not get sent to this device.
Extension Mobility	For Extension Mobility, the device profile settings include DND incoming call alert and DND status. When a user logs in and enables DND, the DND incoming call alert and DND status settings get saved, and these settings get used when the user logs in again. **Note** When a user who is logged in to Extension Mobility modifies the DND incoming call alert or DND status settings, this action does not affect the actual device settings.

Restrictions

Some restrictions apply to DND usage, depending on the phone or device type in use.

- The following phone models and devices that are running SCCP support only the DND Ringer Off option:
  - Cisco Unified IP Phone 7940
  - Cisco Unified IP Phone 7960
  - Cisco IP Communicator

Note Cisco Unified IP Phones 7940 and 7960 that run SIP use their own implementation of Do Not Disturb, which is backward compatible.

- The following phone models and devices support only the DND Call Reject option:
  - Mobile devices (dual mode)
  - Remote Destination Profile
  - Cisco Unified Mobile Communicator

Do Not Disturb Troubleshooting

This section provides troubleshooting information for Cisco Unified IP Phones (SCCP and SIP).

For SIP phones, use the following information for troubleshooting:

- debugs: sip-dnd, sip-messages, dnd-settings
- show: config, dnd-settings
- sniffer traces

For SCCP phones, use the following information for troubleshooting:

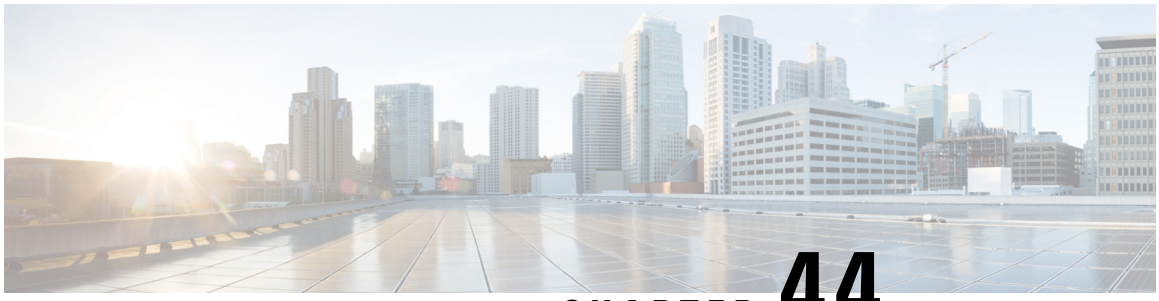
- debug: jvm all info
- sniffer traces

Troubleshooting Errors

The following table describes how to troubleshoot errors with Do No Disturb.

Symptom	Actions
DND softkey does not display or DND feature button does not display	• Verify that the softkey or button template for this phone includes DND. • Capture a sniffer trace and verify that the phone gets the correct softkey or button template. • Verify that the phone firmware is Version 8.3(1) or later.
BLF speed dial does not show DND status	• Verify that the BLF DND is set to enabled in Enterprise parameters. • Capture a sniffer trace and verify that the phone gets the correct notification message. • Verify that the phone firmware is Version 8.3(1) or later.

Symptom	Actions
DND changes are not reflected on the monitoring device.	• Check if the BOT/TCT devices are shared line devices with the DND state set to **OFF**. If the status is set to **ON**, changes to the DND status on other shared lines will not be reflected. • Make sure the DND status on the BOT/TCT devices is set to **OFF** to reflect changes in the DND status on the line you want to monitor.



CHAPTER 44

Privacy

- Privacy Overview, on page 575
- Privacy Configuration Task Flow, on page 576
- Privacy Restrictions, on page 579

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

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the Privacy feature.
Step 2	Enable Privacy Cluster-wide, on page 576	Enable Privacy by default for all the phones in the cluster.
Step 3	Enable Privacy for a Device, on page 576	Enable Privacy for specific devices.
Step 4	Configure Privacy Phone Button Template, on page 577	Configure Privacy phone button template for a device.
Step 5	Associate Privacy Phone Button Template with a Phone, on page 578	Associate the phone button template with a user.
Step 6	Configure Shared Line Appearance, on page 578	Configure the shared line appearance.
Step 7	(Optional) Configure Privacy on Hold, on page 579	Configure Privacy on Hold.

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 5.

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**.

Configure Privacy Phone Button Template

Before you begin

Enable Privacy for a Device, on page 576

Procedure

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

Step 2 Click **Find** to display list of supported phone templates.

Step 3 Perform the following steps if you want to create a new phone button template; otherwise, proceed to the next step.

a) Select a default template for the model of phone and click **Copy**.
b) In the **Phone Button Template Information** field, enter a new name for the template.
c) Click **Save**.

Step 4 Perform the following steps if you want to add phone buttons to an existing template.

a) Click **Find** and enter the search criteria.
b) Choose an existing template.

Step 5 From the **Line** drop-down list, choose feature that you want to add to the template.

Step 6 Click **Save**.

Step 7 Perform one of the following tasks:

- Click **Apply Config** if you modified a template that is already associated with devices to restart the devices.
- If you created a new softkey template, associate the template with the devices and then restart them.

Associate Privacy Phone Button Template with a Phone

Before you begin

Configure Privacy Phone Button Template, on page 577

Procedure

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

Step 2 Click **Find** to display the list of configured phones.

Step 3 Choose the phone to which you want to add the phone button template.

Step 4 In the **Phone Button Template** drop-down list, choose the phone button template that contains the new feature button.

Step 5 Click **Save**.
A dialog box is displayed with a message to press **Reset** to update the phone settings.

Configure Shared Line Appearance

Before you begin

Associate Privacy Phone Button Template with a Phone, on page 578

Procedure

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

Step 2 To locate a specific phone, enter search criteria and click **Find**.
A list of phones that match the search criteria is displayed.

Step 3 Choose the phone for which you want to configure shared line appearance.
The **Phone Configuration** window is displayed.

Step 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.

Step 5 Enter the **Directory Number** and choose the **Route Partition** to which the directory number belongs.

Step 6 Configure the remaining fields in the **Directory Number Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 7 Repeat Step 3, on page 578 to Step 6, on page 578 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

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.

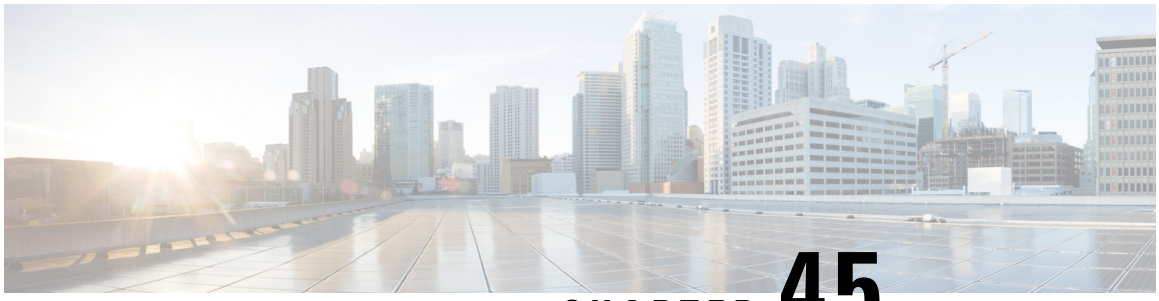
Step 3 From the **Service** drop-down list, choose **Cisco CallManager**.

Step 4 Set the **Enforce Privacy Setting on Held Calls** service parameter to **True**.

Step 5 Click **Save**.

Privacy Restrictions

Restriction	Description
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.



CHAPTER 45

Private Line Automatic Ringdown

- Private Line Automatic Ringdown Overview, on page 581
- Private Line Automatic Ringdown Configuration Task Flow for SCCP Phones, on page 581
- Private Line Automatic Ringdown Configuration Task Flow for SIP Phones, on page 584
- Private Line Automatic Ringdown Troubleshooting, on page 585

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

	Command or Action	Purpose
Step 1	Create Partition, on page 582	Create a partition for the PLAR destination. The only directory number that you can assign to this partition is the PLAR destination.
Step 2	Assign Partitions to Calling Search Spaces, on page 582	Assign the partition to a unique CSS, and a CSS that includes the PLAR destination device.
Step 3	Assign Partition to the Private Line Automatic Ringdown Destination, on page 583	Assign the null partition and a CSS to your PLAR destination directory number.

	Command or Action	Purpose
Step 4	Configure Translation Pattern for Private Line Automatic Ringdown on Phones, on page 583	Create a null translation pattern and assign it to your PLAR destination directory number.

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 From 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

Create Partition, on page 582

Procedure

Step 1 From 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, assign a null partition to the directory number that you want to use as the PLAR destination.

Note Each PLAR destination directory number must have its own unique partition. Do not add any other directory numbers to the null partition that you created for the PLAR destination.

Before you begin

Assign Partitions to Calling Search Spaces, on page 582

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, select the CSS that includes both the null partition and the destination device.
Step 5 Click **Save**.

Configure Translation Pattern for Private Line Automatic Ringdown on Phones

To configure Private Line Automatic Ringdown (PLAR) on phones, configure a null translation pattern and assign the PLAR destination number to that translation pattern.

Before you begin

Assign Partition to the Private Line Automatic Ringdown Destination, on page 583

Procedure

Step 1 In Cisco Unified CM Administration, choose **Call Routing** > **Translation Pattern**.
Step 2 Click **Add New** to create a new translation pattern.
Step 3 Leave the **Translation Pattern** field empty.
Step 4 From the **Partition** drop-down list, select the new partition that you created for the null translation pattern.

Step 5 From the **Calling Search Space** drop-down list, select a calling search space that includes both the new partition and the partition for the PLAR destination device.

Step 6 In the **Called Party Transformation Mask** field, enter the PLAR destination directory number.

Step 7 Click **Save**.

Private Line Automatic Ringdown Configuration Task Flow for SIP Phones

Perform these tasks to configure Private Line Automatic Ringdown (PLAR) on SIP Phones.

Procedure

	Command or Action	Purpose
Step 1	Create SIP Dial Rule for Private Line Automatic Ringdown, on page 584	Create a SIP dial rule for PLAR.
Step 2	Assign Private Line Automatic Ringdown Dial Rule to SIP Phone, on page 585	Assign the PLAR dial rule to the phone.

Create SIP Dial Rule for Private Line Automatic Ringdown

To configure Private Line Automatic Ringdown (PLAR) on SIP phones, you must configure a SIP dial rule for your PLAR destination number.

Before you begin

Create Partition, on page 582

Assign Partitions to Calling Search Spaces, on page 582

Assign Partition to the Private Line Automatic Ringdown Destination, on page 583

Configure Translation Pattern for Private Line Automatic Ringdown on Phones, on page 583

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, 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**.

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 584

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, choose the dial rule that you created for PLAR.
Step 4 Click **Save**.

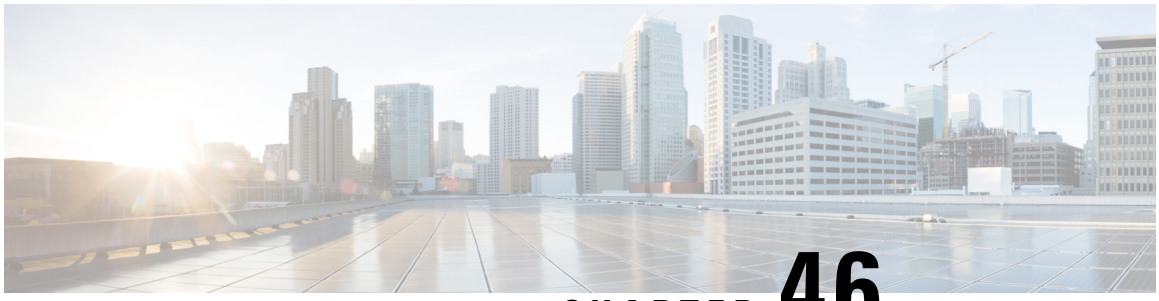
Private Line Automatic Ringdown Troubleshooting

Troubleshooting Private Line Automatic Ringdown on SCCP Phones

Symptom	Solution
The phone goes off hook and the user hears a fast busy (reorder) tone.	Make sure that the CSS that is assigned to the PLAR translation pattern contains the partition of the PLAR destination.
The phone goes off hook and receives dial tone.	Make sure that the CSS that is assigned to the phone contains the partition of the null PLAR translation pattern.

Troubleshooting Private Line Automatic Ringdown on SIP Phones

Symptom	Solution
The phone goes off hook and the user hears fast busy (reorder) tone.	Make sure that the CSS of the SIP phone can reach the PLAR destination.
The phone goes off hook and receives a dial tone.	Make sure that the SIP Dial Rule has been created and is assigned to the phone.



CHAPTER 46

Secure Tone

- Secure Tone Overview, on page 587
- Secure Tone Prerequisites, on page 588
- Secure Tone Configuration Task Flow, on page 588
- Secure Tone Interactions, on page 591
- Secure Tone Restrictions, on page 591

Secure Tone Overview

The Secure Tone feature can configure a phone to play a secure indication tone when a call is encrypted. The tone indicates that the call is protected and that confidential information may be exchanged. The 2-second tone comprises three long beeps. If the call is protected, the tone begins to play on a protected phone as soon as the called party answers.

When the call is not protected, the system plays a nonsecure indication tone, which comprises six short beeps, on a protected phone.

Note Only callers on protected phones can hear secure and nonsecure indication tones. Callers on phones that are not protected cannot hear these tones.

The secure and nonsecure indication tones are supported on the following types of calls:

- Intracluster to IP-to-IP calls
- Intercluster protected calls
- IP-to-Time-Division-Multiplexing (TDM) calls through a protected MGCP E1 PRI gateway

For video calls, the system plays secure and nonsecure indication tones on protected devices.

Note For video calls, the user may first hear secure indication tone for the audio portion of the call and then nonsecure indication tone for overall nonsecure media.

A lock icon that is displayed on a Cisco Unified IP Phone indicates that the media are encrypted, but does not indicate that the phone has been configured as a protected device. However, the lock icon must be present for a protected call to occur.

Protected Device Gateways

You can configure only supported Cisco Unified IP Phones and MGCP E1 PRI gateways as protected devices in Cisco Unified Communications Manager.

Cisco Unified Communications Manager can also direct an MGCP Cisco IOS gateway to play secure and nonsecure indication tones when the system determines the protected status of a call.

Protected devices provide these functions:

- You can configure phones that are running SCCP or SIP as protected devices.
- Protected devices can call nonprotected devices that are either encrypted or nonencrypted. In such cases, the call specifies nonprotected and the system plays nonsecure indication tone to the phones on the call.
- When a protected phone calls another protected phone, but the media is not encrypted, the system plays a nonsecure indication tone to the phones on the call.

Secure Tone Prerequisites

- You must configure the MGCP gateway for SRTP encryption. Configure the gateway with this command: **mgcp package-capability srtp-package**.
- The MGCP gateway must specify an Advanced IP Services or Advanced Enterprise Services image (for example, c3745-adventerprisek9-mz.124-6.T.bin).

Secure Tone Configuration Task Flow

Before you begin

- Review Secure Tone Prerequisites, on page 588

Procedure

	Command or Action	Purpose
Step 1	Generate a Phone Feature List, on page 5	Generate a report to identify devices that support the Secure Tone feature.
Step 2	Configure Phone As a Protected Device, on page 589	Configure the phone as a protected device.
Step 3	Configure Directory Number for Secure Tones, on page 589	Configure multiple calls and call waiting settings for the protected device.
Step 4	Configure Secure Tone Service Parameters, on page 590	Configure service parameters.
Step 5	(Optional) Configure MGCP E1 PRI Gateway, on page 590	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.

Configure Phone As a Protected Device

Before you begin

Generate a Phone Feature List, on page 5

Procedure

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

Step 2 Click the phone for which you want to set secure tone parameters.
The **Phone Configuration** window is displayed.

Step 3 From the **Softkey Template** drop-down list in the Device Information portion of the window, choose **Standard Protected Phone**.

Note
You must use a new softkey template without supplementary service softkeys for a protected phone.

Step 4 Set the **Join Across Lines** option to Off.

Step 5 Check the **Protected Device** check box.

Step 6 From the **Device Security Profile** drop-down list (in the Protocol Specific Information portion of the window), choose a secure phone profile that is already configured in the **Phone Security Profile Configuration** window (**System** > **Security Profile** > **Phone Security Profile**.

Step 7 Click **Save**.

What to do next

Perform one of the following procedures:

- Configure Directory Number for Secure Tones, on page 589
- Configure MGCP E1 PRI Gateway, on page 590

Configure Directory Number for Secure Tones

Before you begin

Configure Phone As a Protected Device, on page 589

Procedure

Step 1 Locate the **Association** section on the **Phone Configuration** window.

Step 2 Select **Add a new DN**.
The **Directory Number Configuration** window is displayed.

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

Step 4 In the **Multiple Call/Call Waiting Settings on Device [device name]** area of the **Directory Number Configuration** window, set the **Maximum Number of Calls** and **Busy Trigger** options to 1.

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

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, choose a server.

Step 3 From the **Service** drop-down list, 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 589

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

Feature	Interaction
Call Transfer, Conference, and Call Waiting	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.
Hold/Resume and Call Forward All	These features are supported on protected calls.

Secure Tone Restrictions

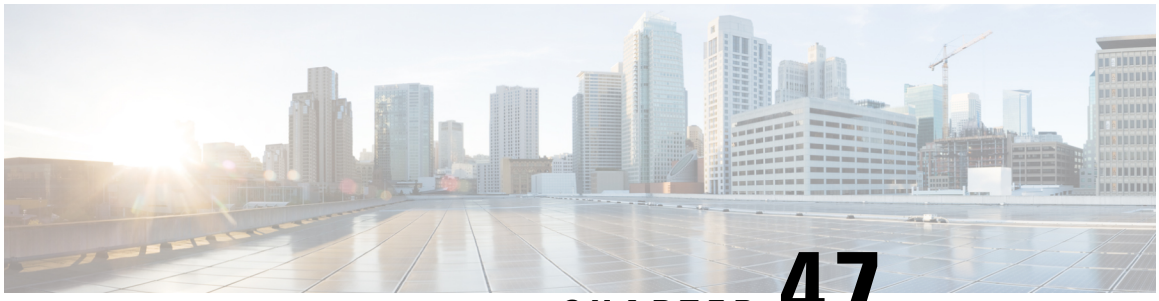
Restriction	Description
Cisco Extension Mobility and Join Across Line services	Cisco Extension Mobility and Join Across Line services are disabled on protected phones.
Shared-line configuration	Shared-line configuration is not available on protected phones.
Non-encrypted media	If the media between the Cisco Unified IP Phone and the MGCP E1 PRI gateway are not encrypted, the call drops.



PART XII

Custom Features

- Branding Customizations , on page 595
- Client Matter Codes and Forced Authorization Codes, on page 603
- Custom Phone Rings and Backgrounds, on page 611
- Music On Hold , on page 619
- Self Care Portal , on page 637
- Emergency Call Handler , on page 641
- Emergency Call Handling with RedSky, on page 655
- Enterprise Groups , on page 661



CHAPTER 47

Branding Customizations

- Branding Overview, on page 595
- Branding Prerequisites, on page 595
- Branding Task Flow, on page 596
- Branding File Requirements, on page 598

Branding Overview

The Branding feature lets you upload customized branding for Cisco Unified Communications Manager. Branding gets applied to the Cisco Unified CM Administration login and configuration windows. Among the items that you can modify include:

- Company logos
- Background colors
- Border colors
- Font colors

Append Logo in Self Care Portal

The Branding feature allows you to append your company logo to the Unified Communications Self Care Portal login page and to the user interface header. You must include the `branding_logo.png` file in your `branding.zip` file and upload the zip file into Cisco Unified Communications Manager. The logo displays in the Self Care Portal after you enable branding in Cisco Unified Communications Manager.

There is no option to customize background colors or fonts for the Self-Care portal.

Branding Prerequisites

You must create your `branding.zip` file that contains the specified folder structure and files. For details, see Branding File Requirements, on page 598.

Branding Task Flow

Complete these tasks to apply branding in Cisco Unified Communications Manager and the Unified Communications Self-Care Portal.

Before you begin

- Review Branding Prerequisites, on page 595

Procedure

	Command or Action	Purpose
Step 1	Configure your branding settings using one of these procedures: • Enable Branding, on page 596 • Disable Branding , on page 597	Apply branding across the Cisco Unified Communications Manager cluster.
Step 2	Restart the Tomcat Service, on page 598	You must restart the Cisco Tomcat service for the new branding setting to get picked up by the Unified Communications Self-Care Portal.

Enable Branding

Use this procedure to enable branding customization for Unified Communications Manager. Branding updates appear even if the system is enabled for SAML Single Sign-On.

Note To enable branding, you must use the primary administrator account with privilege level 4 access. This is the main administrator account that is created during installation.

Note Ensure that you use only one among GUI and CLI to enable branding as well as to disable it. For example, if you enable branding using the GUI interface, you must use the GUI interface itself to disable branding. Else, it will not function properly.

Note Create the `branding.zip` file according to the requirements in the 'Branding File Requirements' section. If the upload is rejected, remove any unsupported or unrelated files, correct the package structure, and upload the file again.

Before you begin

Prepare your `branding.zip` file and save it in a location that Unified Communications Manager can access.

Procedure

Step 1 Log in to Cisco Unified OS Administration.

Step 2 Choose **Software Upgrades** > **Branding**.

Step 3 **Browse** to your remote server and select the `branding.zip` file.

Step 4 Click **Upload File**.

Step 5 Click **Enable Branding**.

Note
You can also enable branding by running the **utils branding enable** CLI command.

Step 6 Refresh your browser.

Step 7 Repeat this procedure on all Cisco Unified Communications Manager cluster nodes.

If you want to append your company logo to the Self-Care Portal user interface, see Restart the Tomcat Service, on page 598.

Disable Branding

Use this procedure to disable branding in your Cisco Unified Communications Manager cluster. You also need to disable branding if you want to remove your company logo from the Self-Care Portal.

Note To disable branding, you must use the primary administrator account with privilege level 4 access. This is the main administrator account that is created during installation.

Note Ensure that you use only one among GUI and CLI to enable branding as well as to disable it. For example, if you enable branding using the GUI interface, you must use the GUI interface itself to disable branding. Else, it will not function properly.

Procedure

Step 1 Log in to Cisco Unified OS Administration.

Step 2 Choose **Software Upgrades** > **Branding**.

Step 3 Click **Disable Branding**.

Note

You can also disable branding by running the **utils branding disable** CLI command.

Step 4 Refresh your browser.

Step 5 Repeat this procedure on all Cisco Unified Communications Manager cluster nodes.

If you want to remove your company logo from the Self-Care Portal user interface, see Restart the Tomcat Service, on page 598

Restart the Tomcat Service

You must restart the Cisco Tomcat service for the branding updates to reflect in the Self-Care Portal.

Before you begin

Make sure that you have completed the following:

- To append your logo to the Self-Care Portal, you must first enable branding in Cisco Unified Communications Manager. The `branding.zip` upload file must include a 44x25 pixel `branding_logo.png` file with your company logo. For details, Enable Branding, on page 596.
- To remove your logo from the Self-Care Portal, you must disable branding in Cisco Unified Communications Manager. For details, Disable Branding , on page 597.

Procedure

Step 1 Log in to the Command Line Interface.

Step 2 Run the **utils service restart Cisco Tomcat** CLI command.

Step 3 Repeat this procedure on all Cisco Unified Communications Manager cluster nodes.

What to do next

After the service restarts, refresh your browser to see the changes in the Self-Care Portal.

Branding File Requirements

Before you apply customized branding to your system, create your `branding.zip` file according to the prescribed specifications. On a remote server, create a `Branding` folder and fill the folder with the specified contents. Once you have added all the image files and subfolders, zip the entire folder and save the file as `branding.zip`.

There are two options for the folder structure, depending on whether you want to use a single image for the header or a combination of six images in order to create a graded effect for the header.

Important Unified CM validates the contents of the `branding.zip` file during upload. The archive must contain only the supported branding folders and files described in this section and must follow the prescribed folder structure. Unified CM rejects a branding package that contains unsupported content or an invalid folder structure.

Table 52: Folder Structure Options

Branding Option	Folder Structure

Single Header Option	If you want a single image for the header background (callout item 3), your branding folder must contain the following subfolders and image files:
	
	`Branding (folder)`
	`ccmadmin (folder)`
	`BrandingProperties.properties (properties file)`
	`brandingHeader.gif (2048*1 pixel image)`
	`ciscoLogo12pxMargin.gif (44*44 pixel image)`
	`branding_logo.png (44*25 pixel image)`

Graded Header Option	If you want to create a graded image for the header background, you need six separate image files to create the graded effect. Your branding folder must contain these subfolders and files
	
	`Branding(folder)`
	`ccmadmin (folder)`
	`BrandingProperties.properties (file)`
	`brandingHeaderBegLTR.gif (652*1 pixel image)`
	`brandingHeaderBegRTR.gif (652*1 pixel image)`
	`brandingHeaderEndLTR.gif (652*1 pixel image)`
	`brandingHeaderEndRTR.gif (652*1 pixel image)`
	`brandingHeaderMidLTR.gif (652*1 pixel image)`
	`brandingHeaderMidRTR.gif (652*1 pixel image)`
	`ciscoLogo12pxMargin.gif (44*44 pixel image)`
	`branding_logo.png (44*25 pixel image)`

User Interface Branding Options

The following images display the customization options for the Cisco Unified CM Administration user interface:

Figure 10: Branding Options for Unified CM Administration Login Screen

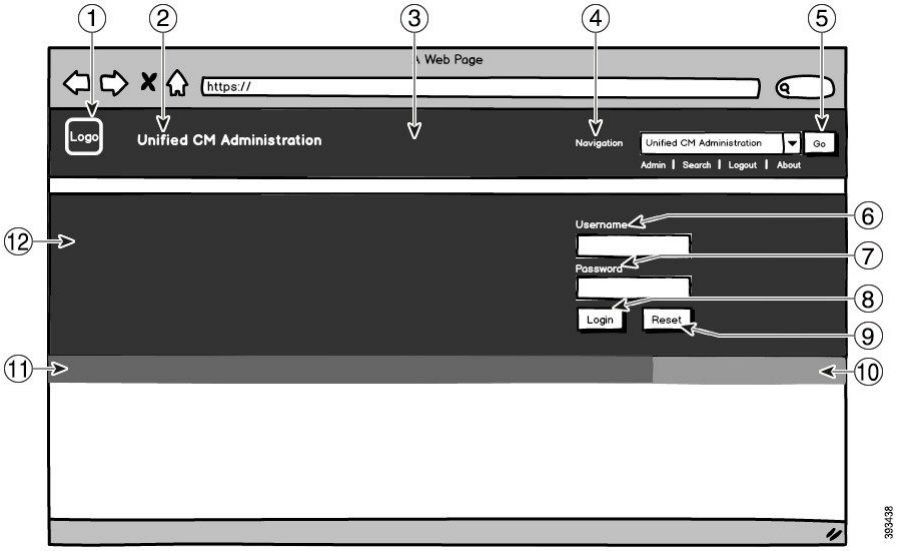
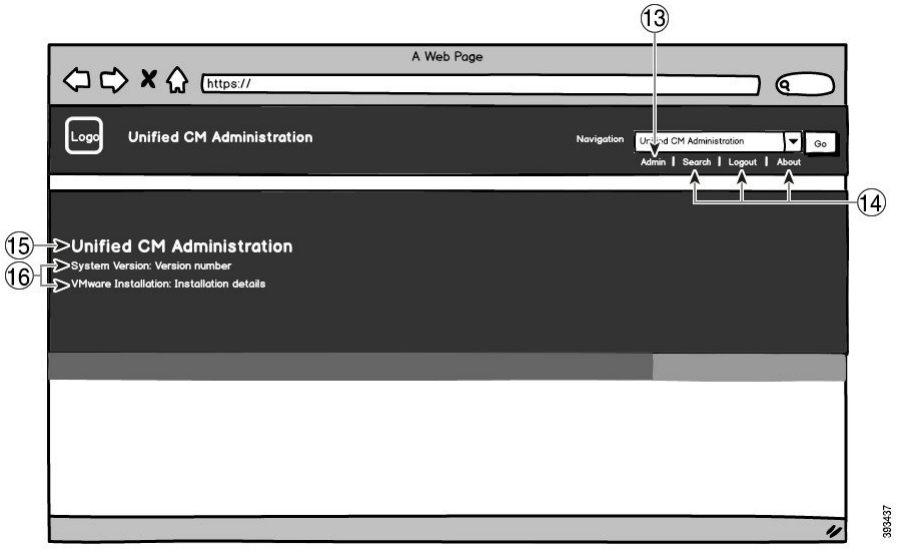


Figure 11: Branding Options for Unified CM Administration Logged In Screen



The following table describes the callout options.

Table 53: User Interface Branding Options: Login Screen

Item	Description	Branding edits
1	Company Logo	To add your logo to Cisco Unified Communications Manager, save your company logo as a 44x44 pixel image with the following filename: `ciscoLogo12pxMargin.gif` (44*44 pixels) **Note** If you also want to append your logo to the header and login screen of the Self-Care Portal, you should also save your logo as the 44x25 pixel `branding_logo.png` file.
2	Unified CM Administration header font color	heading.heading.color
3	Header Background	You can use a single image or a combination of six images to create a grading effect. **Single Image option**—Save your header background as a single image: • brandingHeader.gif (2048*1 pixel) **Graded background option:**—Save your header background as six images for a graded effect: • brandingHeaderBegLTR.gif (652*1 pixel) • brandingHeaderBegRTR.gif (652*1 pixel) • brandingHeaderEndLTR.gif (652*1 pixel) • brandingHeaderEndRTR.gif (652*1 pixel) • brandingHeaderMidLTR.gif (652*1 pixel) • brandingHeaderMidRTR.gif (652*1 pixel)
4	Navigation text	header.navigation.color
5	Go button	header.go.font.color header.go.background.color header.go.border.color
6	Username text	splash.username.color
7	Password text	splash.password.color
8	Login button	splash.login.text.color splash.login.back.ground.color

Item	Description	Branding edits
9	Reset button	splash.reset.text.color splash.reset.back.ground.color
10	Bottom background color – right	splash.hex.code.3
11	Bottom background color – left	splash.hex.code.2
12	Banner	splash.hex.code.1

Table 54: User Interface Branding Options: Logged In Screen

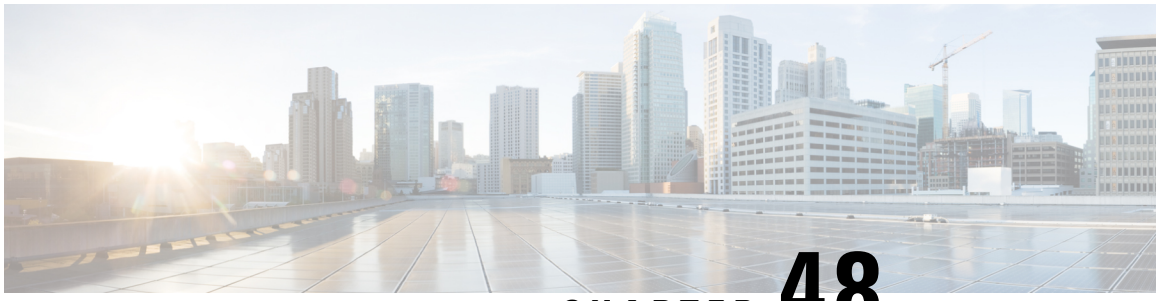
Item	Description	Branding edits
13	User text (for example, 'admin')	header.admin.color
14	Search, About and Login text	header.hover.link.color
15	Unified CM Administration text heading	splash.header.color
16	System Version, VMware Installation text	splash.reset.text.color splash.version.color

Branding Properties Editing Example

Branding properties can be edited by adding the hex code in the properties file (`BrandingProperties.properties`). The properties file uses HTML-based hex code. For example, if you want to change the color of the Navigation text item (callout item #4) to red, add the following code to your properties file:

```
header.navigation.color="#FF0000"
```

In this code, `header.navigation.color` is the branding property that you want to edit, and `"#FF0000"` is the new setting (red).



CHAPTER 48

Client Matter Codes and Forced Authorization Codes

- Client Matter Codes and Forced Authorization Codes Overview, on page 603
- Client Matter Codes and Forced Authorization Codes Prerequisites, on page 603
- Client Matter Codes and Forced Authorization Codes Configuration Task Flow, on page 604
- Client Matter Codes and Forced Authorization Codes Interactions, on page 607
- Client Matter Codes and Forced Authorization Codes Restrictions, on page 608

Client Matter Codes and Forced Authorization Codes Overview

With client matter codes (CMCs) and forced authorization codes (FACs), you can effectively manage call access and accounting. CMCs assist with call accounting and billing for clients, and FACs regulate the types of calls that certain users can place.

CMCs force the user to enter a code; this action specifies that the call relates to a specific client matter. You can assign client matter codes to customers, students, or other populations for call accounting and billing purposes. FACs force the user to enter a valid authorization code that is assigned at a certain access level before the call is completed.

Client Matter Codes and Forced Authorization Codes Prerequisites

- Cisco Unified IP Phones that are running SCCP and SIP support CMC and FAC.
- The CMC and FAC tones play only on Cisco Unified IP Phones that are running SCCP or SIP; TAPI/JTAPI ports; and MGCP FXS ports.

Client Matter Codes and Forced Authorization Codes Configuration Task Flow

You can implement CMCs and FACs separately or together. For example, you may authorize users to place certain classes of calls, such as long distance calls, and also assign the class of calls to a specific client. CMC and FAC tones sound the same to the user; if you configure both codes, the feature prompts the user to enter the FAC after the first tone and enter the CMC after the second tone.

Before you begin

- Review Client Matter Codes and Forced Authorization Codes Prerequisites, on page 603

Procedure

	Command or Action	Purpose
Step 1	To Configure Client Matter Codes, on page 604, complete the following subtasks: • Add Client Matter Codes, on page 605 • Enable Client Matter Codes, on page 605	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.
Step 2	To Configure Forced Authorization Codes, on page 606, complete the following subtasks: • Add Forced Authorization Codes, on page 606 • Enable Forced Authorization Codes, on page 606	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

	Command or Action	Purpose
Step 1	Add Client Matter Codes, on page 605	Determine unique client matter codes that you want to use and add them to your system. Because the number of CMCs directly affects the time that is required for your system to start up, limit the number of CMCs to a maximum of 60,000. If you configure more CMCs than the maximum number, expect significant delays.
Step 2	Enable Client Matter Codes, on page 605	Enable client matter codes through a route pattern.

Add Client Matter Codes

Determine unique client matter codes that you want to use and add them to your system. Because the number of CMCs directly affects the time that is required for your system to start up, limit the number of CMCs to a maximum of 60,000. If you configure more CMCs than the maximum number, expect significant delays.

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 identify the client matter code.

Step 5 Click **Save**.

Enable Client Matter Codes

Enable client matter codes through a route pattern.

Before you begin

Add Client Matter Codes, on page 605

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

	Command or Action	Purpose
Step 1	Add Forced Authorization Codes, on page 606	Determine unique forced authorization codes that you want to use and add them to your system.
Step 2	Enable Forced Authorization Codes, on page 606	Enable forced authorization codes through a route pattern.

Add Forced Authorization Codes

Use this procedure to determine unique forced authorization codes that you want to use and add them to your system. To successfully route a call, the user authorization level must be equal to or greater than the authorization level that is specified for the route pattern for the call.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Forced Authorization Codes**.

Step 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.

Step 3 In the **Authorization Code** field, enter a unique authorization code that is no more than 16 digits.

Users enter this code when they place a call through an FAC-enabled route pattern.

Step 4 In the **Authorization Level** field, enter a three-digit authorization level in the range of 0 to 255.

Step 5 Click **Save**.

Enable Forced Authorization Codes

Use this procedure to enable forced authorization codes through a route pattern.

Procedure

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

Step 2 Perform one of the following tasks:

- Click **Find**, and then choose a route pattern from the resulting list to update an existing route pattern.
- Click **Add New** to create a new route pattern.

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

Table 55: Client Matter Codes and Forced Authorization Codes Interactions

Feature	Interaction
CDR Analysis and Reporting (CAR)	CDR Analysis and Reporting (CAR) allows you to run reports that provide call details for client matter codes (CMCs), forced authorization codes (FACs), and authorization levels.
CTI, JTAPI, and TAPI applications	In most cases, your system can alert a CTI, JTAPI, or TAPI application that the user must enter a code during a call. When a user places a call, creates an ad hoc conference, or performs a consult transfer through a CMC- or FAC-enabled route pattern, the user must enter a code after receiving the tone. When a user redirects or blind transfers a call through a CMC- or FAC-enabled route pattern, the user receives no tone, so the application must send the codes to Cisco Unified Communications Manager. If your system receives the appropriate codes, the call connects to the intended party. If your system does not receive the appropriate codes, Cisco Unified Communications Manager sends an error to the application that indicates which code is missing.
Cisco Web Dialer	Web Dialer supports CMCs and FACs in the following ways: • A user can enter the destination number in the dial text box of the WD HTML page or SOAP request, and then manually enter the CMC or FAC on the phone. • A user can enter the destination number followed by the FAC or CMC in the dial text box of the WD HTML page or SOAP request. For example, if the destination number is 5555, the FAC is 111, and the CMC is 222, a user can make a call by dialing 5555111# (FAC), 5555222# (CMC), or 5555111222# (CMC and FAC). **Note** • WebDialer does not handle any validation for the destination number. The phone handles the required validation. • If a user does not provide a code or provides the wrong code, the call will fail. • If a user makes a call from the WebApp with a DN that contains special characters, the call goes successfully after stripping the special characters. The same rules do not work in SOAP UI.

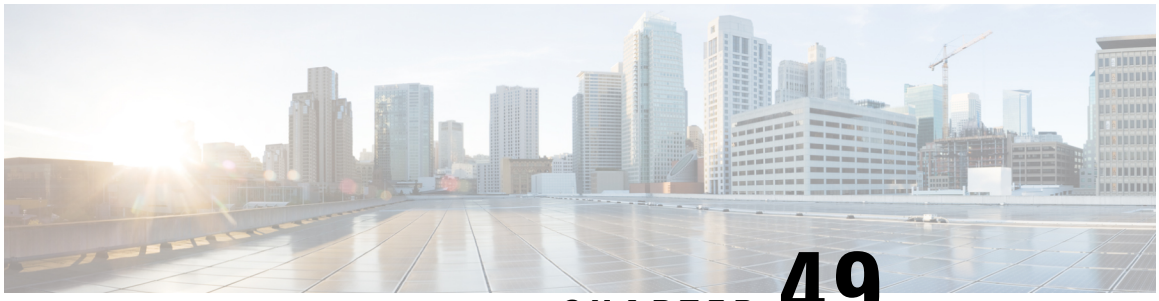
Feature	Interaction
Speed Dial and Abbreviated Speed Dial	You can use speed dial to reach destinations that require a FAC, CMC, dialing pauses, or additional digits (such as a user extension, a meeting access code, or a voicemail password). When the user presses the configured speed dial, the phone establishes the call to the destination number and sends the specified FAC, CMC, and additional digits with dialing pauses inserted.

Client Matter Codes and Forced Authorization Codes Restrictions

Table 56: Client Matter Codes and Forced Authorization Codes Restrictions

Restriction	Description
Analog gateways	H.323 analog gateways do not support CMCs or FACs because these gateways cannot play tones.
Call forwarding	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.
Cisco Unified Mobility	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.
Dial via Office callback number	The CMC and FAC feature on Cisco Mobility does not support an alternative number as its dial via office (DVO) callback number. The DVO callback number must be the number that is registered on the **Mobility Identity** window.
Failover calls	CMCs and FACs do not work with failover calls.
Hearing-impaired users	After dialing the phone number, hearing-impaired users should wait one or two seconds before entering the authorization or client matter code.
Localization	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. **Note** For Cisco Mobility, CMCs and FACs are localized.

Restriction	Description
Overlap sending	The CMC and FAC features do not support overlap sending because Cisco Unified Communications Manager cannot determine when to prompt the user for the code. If you check the **Require Forced Authorization Code** or the **Require Client Matter Code** check box in the **Route Pattern Configuration** window, the **Allow Overlap Sending** check box is automatically unchecked and vice-versa.
Speed-dial buttons	You cannot configure CMCs or FACs for speed-dial buttons. You must enter the code when the system prompts you to do so.



CHAPTER 49

Custom Phone Rings and Backgrounds

- Custom Phone Rings Overview, on page 611
- Custom Phone Rings Prerequisites, on page 611
- Custom Phone Rings Configuration Task Flow, on page 612
- Custom Backgrounds, on page 614
- Custom Backgrounds Configuration Task Flow, on page 615

Custom Phone Rings Overview

Custom Phone Rings allows you to create customized phone rings and upload the customized files to the Cisco Unified Communications Manager TFTP server where they can be accessed by Cisco Unified IP Phones.

Cisco Unified IP Phones ship with default ring types that are implemented in hardware: Chirp1 and Chirp2. In addition, Cisco Unified Communications Manager provides the capability of uploading the following files to phones:

PCM Files

Cisco Unified Communications Manager provides a default set of phone ring sounds that are implemented in software as pulse code modulation (PCM) audio files. Each PCM file specifies a single ring type.

Ringlist.xml File

The Ringlist.xml file describes the list of ring options that are available for phones.

You can upload customized PCM audio files, such as custom ring tones and 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 613.
- The Ringlist.xml file must meet a set of formatting guidelines. For details, review the topic Ringlist.xml File Format Requirements, on page 613.

Custom Phone Rings Configuration Task Flow

Before you begin

- Review Custom Phone Rings Prerequisites, on page 611

Procedure

	Command or Action	Purpose
Step 1	Prepare Custom Phone Rings for Upload, on page 612	Create your customized PCM and Ringlist.xml files.
Step 2	Upload Custom Phone Rings to TFTP Server, on page 612	Upload customized files to the Cisco Unified Communications Manager TFTP server.
Step 3	Restart TFTP Service, on page 613	After the upload completes, restart the Cisco TFTP service.

Prepare Custom Phone Rings for Upload

Procedure

Step 1 Use the `file get tftp <tftp path>` CLI command to download the existing Ringlist.xml file, in addition to any PCM files that you want to modify.

Step 2 Create a PCM file for each ring type that you want to upload. For guidelines on PCM file compatibility with Cisco Unified Communications Manager, see PCM File Format Requirements, on page 613.

Step 3 Use an ASCII editor to update the Ringlist.xml file with your new phone rings. For details on Ringlist.xml file formatting requirements, see Ringlist.xml File Format Requirements, on page 613.

Upload Custom Phone Rings to TFTP Server

Before you begin

Prepare Custom Phone Rings for Upload, on page 612

Procedure

Step 1 From 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**.

Restart TFTP Service

Before you begin

Upload Custom Phone Rings to TFTP Server, on page 612

Procedure

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

Step 2 From the **Server** drop-down list, 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:

```
<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:

```
<CiscoIPPhoneRinglist>   <Ring>
      <DisplayName>Analog Synth 1</DisplayName>
      <FileName>Analog1.raw</FileName>
   </Ring>
   <Ring>
      <DisplayName>Analog Synth 2</DisplayName>
      <FileName>Analog2.raw</FileName>
   </Ring>
</CiscoIPPhoneRinglist>
```

Tip You must include the required DisplayName and FileName for each phone ring type. The Ringlist.xml file can include up to 50 ring types.

Custom Backgrounds

You can also use the TFTP server to upload new custom background images to the phones in your network. Phone users can select the images that you upload as their phone backgrounds. You can configure your system so that phone users can select from an assortment of images or you can assign a specific background image for all phone.

If you want your phone users to be able to customize their phone backgrounds, then you must prepare and upload the following files to the TFTP server whenever you upload new images:

- Full-size background image—Refer to your phone documentation for image specifications, including the image size (in pixels) and color-type, for your phone model.
- A thumbnail image—This is only required if you want your phone users to be able to choose their own background image. Refer to your phone documentation for the thumbnail image specifications
- An edited `List.xml` file—This file contains a listing of the background images from which phone users can select. You must add your new images to this file.

If you want to assign a specific image for all phones then you need to upload the main background image only. In addition, you also must update the Common Phone Profile to direct the phones to use the image that you assign.

Custom Backgrounds Configuration Task Flow

Complete these tasks to configure and upload customized background images for the phones in your deployment. You can configure the system so that phone users can select from an assortment of images, or you can assign a specific background image that displays on all phones.

Procedure

	Command or Action	Purpose
Step 1	Create Phone Background Images, on page 616	Create your full-size background image and corresponding thumbnail image (if required). Refer to your phone documentation for image specifications, including file type, image size (in pixels) and color-type. **Note** The thumbnail is not required if you are assigning a specific background image.
Step 2	Edit the List.xml file, on page 616	Update the `List.xml` file from the appropriate TFTP directory with your new images. This is required so that phone users see the new images in their list of phone background options. **Note** This procedure is required only if you are giving your users the option to choose their own background. If you are assigning a specific background image then there is no need to edit this file.
Step 3	Upload Backgrounds to TFTP Server, on page 617	Upload your files to the TFTP server.
Step 4	Restart the TFTP Server, on page 617	Restart the Cisco TFTP service in order to push the images to your phones.
Step 5	Assign Phone Background for Phone Users, on page 618	Optional. By default, Cisco Unified Communications Manager gives phone users the option to select their own phone background image. However, you can use the Common Phone Profile to assign a specific background image for all phones that use this Common Phone Profile.

Create Phone Background Images

Refer to your phone documentation for background image specifications and thumbnail image specifications. This includes the image sizes (in pixels), file type, and the appropriate destination TFTP directory for that phone model (the TFTP directory is based on the image specifications).

- If you want phone users to have the option to use, or not use, the image that you upload, you must prepare both a full-size image and a thumbnail image according to the specifications for that particular phone model.
- If you want to assign the image to specific phones, you need the full-size image only.

What to do next

If you want phone users to be able to choose their own background image, Edit the List.xml file, on page 616.

If you want to assign a specific background image, you don't need to update the `List.xml` file. Proceed to Upload Backgrounds to TFTP Server, on page 617

Edit the List.xml file

If you want phone users to be able to choose their background images, use this procedure to add any new background images that you upload to the existing `List.xml` file. Each TFTP image directory contains a `List.xml` file that gets used by the phones that use that TFTP directory. This file points to the specific background and thumbnail image for each background option and can include up to 50 background images. The images are listed using the order in which they appear on the phone. For each image, the file contains an `<ImageItem>` element that includes these two attributes:

- Image: Uniform resource identifier (URI) that specifies where the phone obtains the thumbnail image that will appear on the Background Images menu of a phone.
- URL: URI that specifies where the phone obtains the full size image.

Example:

The following example (for a Cisco Unified IP Phone 7971G-GE and 7970G) shows a `List.xml` file that defines two images. The required Image and URL attributes must be included for each image. The TFTP URI that displays in the example is the only supported method for linking to full size and thumbnail images as HTTP URL support is not provided.

```
<CiscoIPPhoneImageList>
   <ImageItem Image="TFTP:Desktops/320x212x12/TN-Fountain.png"
URL="TFTP:Desktops/320x212x12/Fountain.png"/>
   <ImageItem Image="TFTP:Desktops/320x212x12/TN-FullMoon.png"
URL="TFTP:Desktops/320x212x12/FullMoon.png"/>
</CiscoIPPhoneImageList
```

Procedure

Step 1 Log in to the Command Line Interface

Step 2 Run the `file get tftp <filename>` CLI command where <filename> represents the file and filepath of the `List.xml` file for the appropriate TFTP directory.

Note
Make sure that you download the `List.xml` file from the appropriate TFTP directory as each image directory has its own file. Refer to your phone documentation for information on the appropriate TFTP directory for that phone model as the directory is based on the image specifications.

Step 3 Edit the xml file with a new `<ImageItem>` element for each new background option that you want to add.

Upload Backgrounds to TFTP Server

Use this procedure to upload new phone background files to the TFTP server.

- If you want your phone users to be able to choose their own background image, then you must upload your full-size background image, a thumbnail image, and the updated `List.xml` file.
- If you are assigning a specific background image, you need to upload the full-size background image only.

Procedure

Step 1 From Cisco Unified OS Administration, choose **Software Upgrades** > **TFTP File Management**

Step 2 Click **Upload File** and do the following:

a) Click **Choose File** and select the background file that you want to upload.
b) In the **Directory** field, enter the appropriate TFTP directory for that phone model. The TFTP directory corresponds to the image size and color type. Refer to your phone documentation for images specification.
c) Click **Upload File**
d) Repeat these steps to upload both the thumbnail image and `list.xml` files as well. These files should be loaded to the same TFTP directory as the main background image.

Step 3 Click **Close**.

Restart the TFTP Server

After you have uploaded your custom files to the TFTP directory, restart the Cisco TFTP server to push the files to the phones.

Procedure

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

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

Assign Phone Background for Phone Users

By default, Cisco Unified Communications Manager allows phone users to customize their own phone background image. However, you can use the Common Phone Profile setting to assign a specific background image for all phones that use this Common Phone Profile.

Procedure

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

Step 2 Do one of the following:

- Click **Find** and select the Common Phone Profile that your phones use.
- Click **Add New** to create a new Common Phone Profile.

Step 3 If you want users to be able to choose their background image, make sure that the **Enable End User Access to Phone Background Image Setting** check box is checked (this is the default setting).

Step 4 If you want to assign a specific background image for phones that use this profile:

- Uncheck the **Enable End User Access to Phone Background Image Setting** check box.
- In the **Background Image** text box, enter the filename of the image file that you want to assign. Also, check the **Override Enterprise Settings** check box that corresponds to this text box.

Step 5 Complete any remaining fields in the **Common Phone Profile** window. For help with the fields and their settings, refer to the online help.

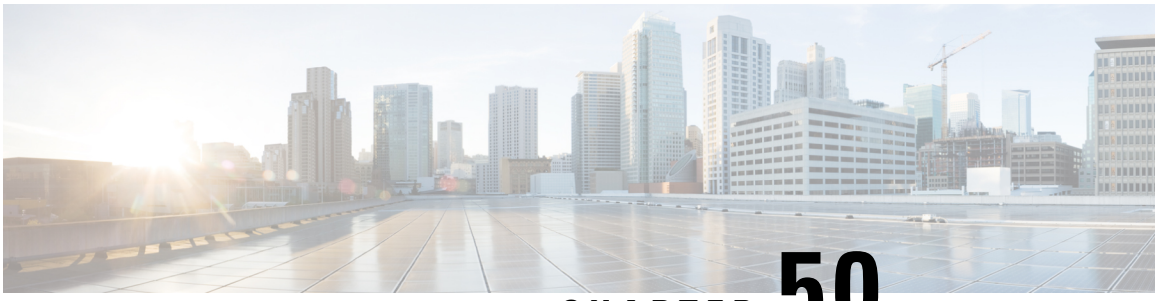
Step 6 Click **Save**.
If you have assigned a specific background image, all phones that use this Common Phone Profile will use the image that you specify.

What to do next

If you have created a new Common Phone Profile, reconfigure your phones so that they use this profile. For details on how to configure phones in Cisco Unified Communications Manager, see the "Configure Endpoint Devices" section of the *System Configuration Guide for Cisco Unified Communications Manager*.

Tip If you have a large number of phones to assign, use the Bulk Administration Tool to assign a Common Phone Profile to a large number of phones in a single operation. For details, see the *Bulk Administration Guide for Cisco Unified Communications Manager*.

Note After you complete your configuration, reset your phones.



CHAPTER 50

Music On Hold

- Music On Hold Overview, on page 619
- Interwork External Multicast MOH to Unicast MOH, on page 623
- Music On Hold Prerequisites, on page 624
- Music On Hold Configuration Task Flow, on page 624
- Unicast and Multicast Audio Sources, on page 631
- Music On Hold Interactions, on page 633
- Music On Hold Restrictions, on page 634
- Music On Hold Troubleshooting, on page 636

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).

For details, see the Cisco Unified Customer Voice Portal documentation at http://www.cisco.com/c/en/us/support/customer-collaboration/unified-customer-voice-portal/tsd-products-support-series-home.html.

Increased Capacity of IP Voice Media Streaming Application and Expanded MOH Audio Source

Cisco IP Voice Media Streaming application is installed automatically when you install Cisco Unified Communications Manager. Activate this application to enable the Music On Hold (MOH) feature.

With this release, the capacity of Cisco Unified Communications Manager to support unique and concurrent MOH audio sources, while the Music On Hold service is running on the MOH server, is increased from 51 to 501. The MOH audio sources are numbered from 1 to 501 with the fixed MOH audio source remaining at the number 51.

The fixed MOH device cannot use an audio source that connects through a USB MOH device, because Cisco Unified Communications Manager does not support USB when running on VMware. Use of the fixed MOH USB device is not supported on VMware. However, provision the external sound device for use with deployments that utilize Cisco Unified Survivable Remote Site Telephony (SRST) multicast MOH.

You can configure each MOH audio source to use a custom announcement as an initial greeting and/or an announcement that is played periodically to callers who are hearing the music. Cisco Unified Communications Manager provides 500 custom announcements that you can use on one or multiple MOH audio sources. These announcements are not distributed between the Cisco Unified Communications Manager servers within a cluster. You have to upload these custom announcement files to each server that provides the MOH and announcement services. You must also upload each custom music file for MOH audio sources to each server.

Performance Impact of Media Devices with Services

The Cisco IP Voice Media Streaming application runs as a service for four media devices—annunciator (ANN), software conference bridge, Music On Hold (MOH), and software media termination point. Activate this service on a Cisco Unified Communications Manager server as coresident with call processing. When you activate this service, ensure that you configure these media devices for limited capacity to avoid any impact on the call processing. The default settings for the media devices are defined based on this coresident operation. You can adjust these settings by reducing the use of one or more media devices to increase other settings.

For example, if you are not using software media termination point devices, you can choose the **Run Flag** setting for the SW MTP to **False**, select **System** > **Service Parameters** > **Cisco IP Voice Media Streaming App service** > **MTP Parameters** , and add the **MTP Call Count** setting to **Media Resource** > **MOH Server** > **Maximum Half Duplex Streams** configuration. Depending on the call traffic, you can modify the default settings. However, monitor the server performance activity for CPU, memory, and IO wait. For higher capacity clusters, such as the ones using 7500 user OVA configuration, it is possible to increase the default media device settings for Call Count by 25%.

For installations where you expect high usage of the media devices, such as Music On Hold, or where high call volumes require higher number of media connections, activate the Cisco IP Voice Media Streaming application service on one or more of the Cisco Unified Communications Manager servers which do not have call processing activated. Activating this service limits the impact of media device usage to other services,

such as call processing. Then, you can increase the configuration settings for maximum number of calls for the media devices.

When you activate Cisco IP Voice Media Streaming application as co-resident with Cisco Unified Communications Manager service, it can impact call processing performance. To increase the capacity settings for Music On Hold or annunciator from the default settings, it is suggested to activate Cisco IP Voice Media Streaming application on a server without activating Cisco Unified Communications Manager.

The CPU performance is impacted by MOH when active callers are on hold or when multicast MOH audio streams are configured.

Table 57: General Performance Results

Configuration Notes	CPU Performance
Dedicated MOH server, 1000 held calls, 500 MOH sources with greeting and periodic announcements.	25–45% (7500 user OVA configuration)
Native call queuing with dedicated MOH server and annunciator server, 1000 queued calls, 500 MOH sources with greeting and periodic announcements. An annunciator can play up to 300 simultaneous greeting announcements.	25–45% (7500 user OVA configuration)
Dedicated MOH server, 500 held calls, 500 MOH sources with greeting and periodic announcements.	15–35% (7500 user OVA configuration)

Table 58: Extrapolated Recommendations

Configuration	Recommendation Limit
When Cisco IP Voice Media Streaming application is co-resident with Cisco Unified Communications Manager on 2500 OVA (moderate call processing).	MOH: 500 held callers, 100 MOH sources, and 48 to 64 annunciator callers.
When Cisco IP Voice Media Streaming application is a dedicated server on 2500 OVA.	MOH: 750 held callers, 250 MOH sources, and 250 annunciator callers.
When Cisco IP Voice Media Streaming application is co-resident with Cisco Unified Communications Manager on 7500/10K OVA (moderate call processing).	MOH: 500 held callers, 250 MOH sources, and 128 annunciator callers.
When Cisco IP Voice Media Streaming application is a dedicated server on 7500/10K OVA.	MOH: 1000 held callers, 500 MOH sources, and 300-700 annunciator callers (with 1 MOH codec). **Note** Reduce annunciator to 300 for two MOH codecs.



Note These recommendations are specific to MOH/ANN devices. If you combine these devices with the software media termination point (MTP) and call forward busy (CFB) devices, reduce the limits to provide streams.

Configuration Limitations for Capacity Planning

The Cisco IP Voice Media Streaming application and Self Provisioning IVR services use a media kernel driver to create and control Real-time Transfer Protocol (RTP) streams. This media kernel driver has a capacity of 6000 streams. These streams allow the media devices and IVR to make resource reservations.

These reservations are based on the following capacity calculations:

Media Device	Capacity
Annunciator	(Call Count service parameter) * 3 Where 3 indicates total of receiving (RX) and transmitting (TX) calls for endpoint and 1 for .wav file.
Software Conference Bridge	(Call Count service parameter) * 2 Where 2 indicates total streams of RX and TX endpoints.
Software Media Termination Point	(Call Count service parameter) * 2 Where 2 indicates total streams of RX and TX endpoints.
Music On Hold	((Maximum Half Duplex Streams) * 3) + (501 * 2 * [number of enabled MOH codecs]) Where: • (Maximum Half Duplex Streams) is a configuration setting on the MOH device configuration administration web page. • 3 indicates total steams of RX, TX, and greeting announcement .wav file. • 501 indicates the maximum number of Music On Hold (MOH) sources. • 2 indicates music .wav stream and possible multicast TX stream. • [number of enabled MOH codecs] is based on how many MOH codecs are enabled in the Cisco IP Voice Media Streaming application service parameters.
Self Provisioning IVR Service	(500 * 2) Where 500 indicates callers, and 2 indicates total streams from RX and TX streams.

Hence, to enable MOH to support a maximum of 1000 callers, use the following equation: `1000 * 3 + 501 * 2 * 1 = 4002` driver streams with one enabled codec and `1000 * 3 +501 * 2 * 2 = 5004` with two enabled codecs. Reduce the remaining devices and deactivate the Self Provisioning IVR service to limit total reservations to 6000, which allows the MOH device to make these reservations. It may also require that you do not activate the Self Provisioning IVR service on the same server with Cisco IP Voice Media Streaming application.

If configuration settings of the media devices exceed the capacity of the media device driver, the media devices that register with the device driver first will be able to reserve their required stream resources. The media devices that register later are restricted to fewer than requested stream resources. The later registered media

devices result in logging some alarm messages and automatically reducing the call count for the restricted media device.

Note A media kernel driver with a capacity of 6000 streams might not support that many simultaneous media device connections.

Interwork External Multicast MOH to Unicast MOH

With this release, you can configure a Cisco Unified Survivable Remote Site Telephony (SRST) router as an audio source. This router provides multicast MOH audio for devices that are capable of multicast reception. In this approach, devices act as if Cisco Unified Communications Manager is sending the multicast MOH audio. However, devices that are capable of only the unicast reception cannot hear the MOH audio that an external MOH source (for example, Cisco Unified SRST router) sends. Examples of devices that are capable of unicast reception only can be public switched telephone network (PSTN) phones, destination to session border controllers (SBC), and Session Initiation Protocol (SIP) trunks.

In this release of Cisco Unified Communications Manager, this feature is enhanced to receive multicast MOH audio from an external audio source and send it as unicast MOH audio. Cisco Unified Communications Manager uses this feature to play multicast MOH audio as unicast MOH for the devices that are capable of unicast MOH reception only. Examples of an external MOH audio source can be a Cisco Unified SRST router or software that can send multicast MOH audio.

An administrator configures the fields for this feature from Cisco Unified CM Administration **Music On Hold Audio Source Configuration** window.

Note

- This feature has no impact on existing functionality of playing multicast MOH audio using an external audio source for the devices that are capable of multicast reception.
- For the unicast media connection, Cisco Unified Communications Manager MOH Server plays the initial announcement and periodic announcement even if you configure the MOH audio source with external multicast source.

Configuration Tips for the Codec-Specific Inbound Audio Stream

Configure an external multicast audio source, such as Cisco Unified SRST router, to MOH server for streaming the required audio feed.

To configure an external multicast audio source, such as a Cisco Unified SRST router, configure the **Source IPv4 Multicast Address** and **Source Port Number** fields in the **MOH Audio Source Configuration** window.

- Cisco Unified Communications Manager listens to multicast G.711 mu-law stream on external multicast IP address and port that you configured on the **MOH Audio Source configuration** window. An MOH server can transcode between the G.711 mu-law or a-law or L16 256K wideband MOH codecs. The external multicast RTP stream uses G.711 mu-law codec for MOH as a source for G.711 mu-law or a-law or L16 256K wideband MOH codecs. For G.711 a-law and wideband calls, Cisco Unified Communications Manager MOH server transcodes the inbound G.711 mu-law stream to outbound G.711 a-law or wideband stream before sending it to the device.

- Cisco Unified Communications Manager listens to multicast G.729 stream on external multicast IP and port value added with four that is configured on the **MOH audio source configuration** window. For example, if you configure an MOH audio Source with 239.1.1.1:16384, Cisco Unified Communications Manager listens to G.711 mu-law stream on 239.1.1.1:16384 and G.729 stream on 239.1.1.1:16388 (port value added with four). An MOH server cannot transcode for G.729 codecs. Callers who are using MOH G.729 codec require an external multicast RTP stream using G.729 or G.729a codec.

Music On Hold Prerequisites

- Before you configure multicast, ensure that you configure MOH server and audio sources. If you want to use fixed audio source, configure it before you configure multicast.
- Make sure to decide whether you are going to do unicast or multicast Music On Hold
- It is crucial to plan the capacity of the deployed and configured hardware and ensure that it can support the anticipated call volume of the network. You need to know the hardware capacity for MOH resources and consider the implications of multicast and unicast MOH in relation to this capacity. Ensure that network call volumes do not exceed these limits. When MOH sessions reach these limits, an additional load can result in poor MOH quality, erratic MOH operation, or loss of MOH functionality.
- If you use multicast MOH and the devices that listen to multicast MOH streams are not in the same IP network, you must enable multicast routing in the IP network. Take care when you enable the multicast routing to avoid the potential flooding of parts of the network with wrongly sent multicast packets (specially, across WAN links). Disable multicasts on interfaces on which the multicast MOH packets are not required and use the Max Hops parameter.
- For detailed information on planning your Music On Hold deployment, including server capacities, refer to the Music On Hold capacities topics in the *Cisco Collaboration System Solution Reference Network Design*.

Music On Hold Configuration Task Flow

Complete these tasks to configure Music On Hold (MOH) for your system.

Procedure

	Command or Action	Purpose
Step 1	Activate Cisco IP Voice Media Streaming, on page 625	Activate the **Cisco IP Voice Media Streaming Service Application** service to enable Music On Hold.
Step 2	Configure Music On Hold Server, on page 626	Configure basic server settings for the MOH server.
Step 3	Upload Audio File for Music On Hold, on page 626	Optional. Upload your own audio files to make them available as MOH audio streams.

	Command or Action	Purpose
Step 4	Configure Music On Hold Audio Source, on page 627	Configure Music On Hold audio streams. You can also associate an uploaded audio files to an MOH audio stream.
Step 5	Configure Fixed Music On Hold Audio Source, on page 628	Configure the fixed Music On Hold audio source. The system supports a single fixed MOH audio source (stream 51).
Step 6	Add MOH to Media Resource Group, on page 628	Assign the Music On Hold service to a Media Resource Group. The group compiles the media resources that are available to an endpoint in a call.
Step 7	Configure Media Resource Group List, on page 629	Assign your Media Resouce Groups to a prioritized Media Resource Group List.
Step 8	Add Media Resources to Device Pool, on page 629	Make Music On Hold available to endpoints by assigning the Media Resource Group List to a device or device pool.
Step 9	Configure MOH Service Parameters, on page 630	Optional. Configure optional Music On Hold parameters such as default codecs and default audio streams for calls on hold.

Activate Cisco IP Voice Media Streaming

The **Cisco IP Voice Media Streaming Application** service must be **Activated** in order to use Music On Hold.

Note During installation, Unified Communications Manager installs and configures a default Music On Hold audio source. Music On Hold functionality can proceed by using the default audio source.

Procedure

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

Step 2 Choose a server from the **Server** drop-down list.

Step 3 Under **CM Services** , make sure the **Cisco IP Voice Media Streaming App** service is **Activated**. If the service is deactivated, check the service and click **Save**.

Configure Music On Hold Server

Before you begin

Make sure one or multiple Music On Hold (MOH) servers are available.

Note The Cisco Unified Communications Manager MOH server is automatically added when the **Cisco IP Voice Media Streaming Application** service is activated.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Media Resources** > **Music On Hold Server**.

Step 2 Click **Find** and select the Music On Hold server that you want to update.

Step 3 Select the **Host Server**.

Step 4 Enter a descriptive **Music On Hold Server Name** along with a **Description**.

Step 5 Select the **Device Pool** you want to use for this server.

Step 6 Configure server capacity by configuring the following fields:

- **Maximum Half Duplex Stream**—Set this to the maximum number of devices that can be on unicast music on hold that is streamed from this music on hold server at any given time. You can use the following formula to calculate the maximum:

  Note
  `(Server and deployment capacity) – ([Number of multicast MOH sources] * [Number of enabled MOH codecs])`

- **Maximum Multi-cast Connections**—Set this to a value that is greater than or equal to the number of devices that might be placed on multicast MOH at any given time.

Step 7 (Optional) To enable multi-casting, check the **Enable Multi-cast Audio Sources on this MOH Server** check box, and configure the multicast IP address ranges.

Step 8 Configure the additional fields in the **Music On Hold Server Configuration** window. For help with the fields and their settings, see the online help.

Step 9 Click **Save**.

Upload Audio File for Music On Hold

Use this procedure if you want to upload customized audio files that you can make available as Music On Hold audio streams.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Media Resources** > **MOH Audio File Management**.

Step 2 Click **Upload File**.

Step 3 Click **Choose File** and browse to the file you want to upload. Once you've selected the file, click **Open**.

Step 4 Click **Upload**.

The **Upload Result** window shows the result of the upload. The uploading procedure uploads the file and performs audio conversions to create codec-specific audio files for MOH. Depending on the size of the original file, processing may take several minutes to complete.

Step 5 Click **Close** to close the **Upload Result** window.

Step 6 Repeat this procedure if you want to upload additional audio files.

Note
When you import an audio source file, Unified Communications Manager processes the file and converts the file to the proper formats for use by the Music On Hold server. Following are examples of valid input audio source files:

- 16-bit PCM .wav file
- Stereo or mono
- Sample rates of 48 kHz, 44.1 kHz, 32 kHz, 16 kHz, or 8 kHz

Note
MOH audio source files do not automatically propagate to other MOH servers in a cluster. You must upload an audio source file to each MOH server or each server in a cluster separately

Configure Music On Hold Audio Source

Use this procedure to configure Music On Hold audio sources. You can configure audio streams and associate uploaded files to an audio stream. You can configure up to 500 audio streams.

Note If a new version of an audio source file is available, perform the update procedure to use the new version.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Media Resources** > **Music On Hold Audio Source**.

Step 2 Do either of the following:

- Click **Find** and select an existing audio stream.
- Click **Add New** to configure a new stream.

Step 3 From the **MOH Audio Stream Number**, select an audio stream.

Step 4 Enter a unique name in the **MOH Audio Source Name** field.

Step 5 Optional. Check the **Allow Multi-casting** check box if you want to allow this file to be multi-casted.

Step 6 Configure the audio source:

- Check the **Use MOH WAV file** source radio button and from the **MOH Audio Source File**, select the file you want to assign.
- Check the **Rebroadcast External Multicast Source** radio button and enter the multicast source IP Address details.

Step 7 In the **Announcement Settings for Held and Hunt Pilot Calls** section, assign the announcements that you want to use for this audio source.

Step 8 Configure the remaining fields in the **Music On Hold Audio Source Configuration** window. For help with the fields and their settings, see the online help.

Step 9 Click **Save**.

Configure Fixed Music On Hold Audio Source

For each cluster, you may define one fixed audio source (Source 51). You must set up the fixed audio source that is configured per cluster on each MOH server. The fixed audio source originates from a fixed device that uses the local computer audio driver.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Media Resources** > **Fixed MOH Audio Source**.

Step 2 Optional. Check the **Allow Multi-casting** check box if you want to allow this audio source to be multi-casted.

Step 3 Check the **Enable** check box to enable the fixed audio source. When you check this check box, a **Name** is required.

Step 4 In the **Announcement Settings for Held and Hunt Pilot Calls** area, configure announcements for this audio source.

Step 5 Configure the fields in the **Fixed MOH Audio Source Configuration** window. For help with the fields and their settings, see the online help.

Step 6 Click **Save**.

Add MOH to Media Resource Group

A Media Resource Group is a logical grouping of media resources. You may associate a media resource group with a geographical location or a site, as required. You can also form media resource groups to control server usage, or unicast or multicast service type.

Procedure

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

Step 2 Do either of the following:

- Click **Find** and select an existing group.
- Click **Add New** to create a new group.

Step 3 Enter a **Name** and **Description**.

Step 4 In the **Available Media Resources** list, select the Music On Hold resource and use the down arrow to add the resource to the **Selected Media Resources**. Repeat this step for the other media resources you want to assign to this group.

Step 5 (Optional) Check the **Use Multi-cast for MOH Audio** check box if you want to allow Music On Hold multi-casting.

Step 6 Click **Save**.

Configure Media Resource Group List

Media Resource Group List lists the prioritized media resource groups. An application can select required media resources from among ones that are available according to the priority order that is defined in a media resource group list.

Procedure

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

Step 2 Do either of the following:

- Click **Find** and select an existing media resource group list.
- Click **Add New** to create a new media resource group list.

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

Step 4 From the **Available Media Resource Groups** list, select the groups you want to add to this list and use the down arrow to move them to **Selected Media Resource Groups**.

Step 5 In the **Selected Media Resource Groups** list use the up and down arrows to the right of the list to edit the prioritized order of groups.

Step 6 Click **Save**.

Add Media Resources to Device Pool

You can make MOH available to devices by assigning the media resource group list that contains the MOH resource to a device or to the device pool.

Note The device in a call will use the media resource group list that is assigned to the device in the **Phone Configuration** window. If none is assigned, it will use the media resource group list for the device pool that is used for the call.

Procedure

Step 1 From Cisco Unified CM Administration, do either of the following:

- Choose **System** > **Device Pool**.
- Choose **Device** > **Phone**.

Step 2 Click **Find** and select an existing phone or an existing device pool.

Step 3 From the **Media Resource Group List** drop-down list, select the media resource group list that contains the Music On Hold resource.

Step 4 Complete the remaining fields in the configuration window. For more information on the fields and their configuration options, see Online Help.

Step 5 Click **Save**.

Configure MOH Service Parameters

Use this procedure to configure optional service parameters for Music On Hold (MOH). For many deployments the default settings will be sufficient.

Procedure

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

Step 2 From the **Server** drop-down list, select the server.

Step 3 From the **Service** drop-down list, select **Cisco IP Voice Media Streaming**.

Step 4 From the **Clusterwide Parameters (Parameters that apply to all servers)** area, configure optional MOH service parameters.

Step 5 Click **Save**.

Step 6 From the **Service** drop-down list, select **Cisco CallManager**.

Step 7 Configure optional MOH parameters. For example, under **Clusterwide Parameters (Service)**, you can assign the default audio sources for Hold.

Step 8 Click **Save**.

Note
All parameters apply only to the current server except parameters that are in the cluster-wide group.

View Music on Hold Audio File

View existing Music On Hold audio files that are stored on the system.

Procedure

Step 1 In Cisco Unified CM Administration, choose **Media Resources** > **MOH Audio File Management**. The **Music On Hold Audio File Management** window appears.

Step 2 View the following information for each record:

- **Check box**—If the audio file can be deleted, a check box appears before the **File Name** column.
- **File Name**—This column displays the audio file name.
- **Length**—This column displays the audio file length in minutes and seconds.
- **File Status**—This column displays one of the following statuses of an audio file:
  - **Translation Complete**—This status appears after a file is uploaded successfully and is available for use as audio files for a music on hold audio source.
  - **In Use**—This status appears after you add a Music On Hold audio source that uses this audio file as its MOH audio source file.

  Note
  You cannot a delete a file with **In Use** status.

Unicast and Multicast Audio Sources

Unicast Music On Hold is the system default option. However, you need to configure for multicast, if required. Both multicast and unicast configurations present the same audio-source behavior to held parties. Each audio source is used once, and the stream is split internally and is sent to the held parties. The only difference between multicast and unicast, in this case, is how the data is sent over the network.

Table 59: Differences Between Unicast and Multicast Audio Sources

Unicast Audio Source	Multicast Audio Source
Consists of streams that are sent directly from the MOH server to the endpoint that requests an MOH audio stream.	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.
A unicast MOH stream is a point-to-point, one-way audio RTP stream between the server and the endpoint device.	A multicast MOH stream is a point-to-multipoint, one-way audio RTP stream between the MOH server and the multicast group IP address.
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.	Enables multiple users to use the same audio source stream to provide MOH.

Unicast Audio Source	Multicast Audio Source
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.	For multicast users, this announcement is not heard.
The additional MOH streams can have a negative effect on network throughput and bandwidth.	Multicast MOH conserves system resources and bandwidth.
Extremely useful in networks in which multicast is not enabled or devices are incapable of multicast.	Can be problematic in situations in which a network is not enabled for multicast or the endpoint devices are incapable of processing multicast.
Includes managing devices only.	Includes managing devices, IP addresses, and ports.
No requirement to define the Music On Hold server.	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.
Functions without configuring MOH audio source, MOH server, or media resource group list.	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.

Note The Multicast MOH Direction Attribute for SIP service parameter determines whether Cisco Unified Communications Manager sets the direction attribute of the Session Description Protocol (SDP) in its multicast Music On Hold (MOH) INVITE message to sendOnly or recvOnly.

If your deployment uses SIP phone uses Release 8.4 and earlier for Cisco Unified IP Phones 7940 and 7960, or SIP phone uses Release 8.1(x) and earlier for Cisco Unified IP Phones 7906, 7911, 7941, and 7961, set this parameter to **sendOnly**. Otherwise, leave this parameter set to the default value, **recvOnly**.

Music On Hold Interactions

Feature	Interaction
Multicast Music On Hold over H.323 Intercluster Trunks	Using the multicast MOH over H.323 intercluster trunk feature, you can multicast MOH to work over H.323 intercluster trunks (ICT). When a call connects over an intercluster trunk and one of the parties presses the Hold key, MOH streams over the intercluster trunk. If you have turned on the multicast MOH and have configured the holding party and trunk to use the multicast MOH server, MOH streams with multicast. Only one multicast MOH stream streams over the trunk regardless of the number of calls that are put on hold on this trunk. Additional points regarding this feature: • This feature does not work if any middle box between Cisco Unified Communications Managers does not pass the new fields in Terminal Capability Set (TCS) and OLC message. • This feature requires no additional configuration for field up multicast MOH, and applies only between Cisco Unified Communications Managers that support single-transmitter multicast. • The feature is On by default, but can be turned off by setting the **Send Multicast MOH in H.245 OLC Message** service parameter to **False**. Setting this value can resolve interoperability issues that the feature might cause.
Music On Hold Failover and Fallback	The MOH server supports Cisco Unified Communications Manager lists and failover as implemented by the software conference bridge and media termination point. Upon failover, the system maintains connections to a backup Cisco Unified Communications Manager, if available. When a Music On Hold server fails during an active Music On Hold session, the held party hears no music from this point. However, this situation does not affect normal call functions.
Call Park and Directed Call Park	Music On Hold allows users to place calls on hold with music that a streaming source provides. Music On Hold allows two types of hold: • User hold—The system invokes this type of hold when a user presses the Hold button or Hold softkey. • Network hold—This type of hold takes place when a user activates the Transfer, Conference, or Call Park feature, and the hold automatically gets invoked. This hold type applies to directed call park because directed call park is a transfer function. However, Directed Call Park uses the Cisco Call Manager service parameter, Default Network Hold MOH Audio Source, for the audio source.
Extension Mobility Cross Cluster—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).

Feature	Interaction
Hold Reversion	Cisco Unified Communications Manager supports MOH on a reverted call if MOH is configured for a normal held call.
Media Resource Selection	Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a Music On Hold resource.
Secured Music On Hold with SRTP	Cisco Unified Communications Manager enhances the Cisco IP Voice Media Streaming application service to support Secure Real-Time Protocol (SRTP). Hence, when you enable the Cisco Unified Communications Manager cluster or system for security, the MOH server registers with Cisco Unified Communications Manager as an SRTP capable device. If the receiving device is also SRTP-capable, the music media is encrypted before streaming to the receiving device. Make sure of the following: • Cluster security should be mixed mode—Run the `utils ctl set-cluster mixed-mode` CLI command • SIP trunks in the path support SRTP—The **SRTP Allowed** check box must be checked in the **Trunk Configuration** window for SRTP to work over the trunk. • Devices support SRTP—In the Phone Security Profile used by the endpoint, the **Device Security Mode** must be **Encrypted**.

Music On Hold Restrictions

Restriction	Description
Multicast Music On Hold Support	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.
Internet Protocol Support	Multicast Music On Hold supports only IPv4. The Cisco IP Voice Media Streaming Application, which is a component of Music On Hold, supports both IPv4 and IPv6 audio media connections for unicast Music On Hold. Multicast Music On Hold supports IPv4 only. Devices with an IP addressing mode of IPv6 only cannot support multicast.
Distribution of fixed-device audio sources	Cisco Unified Communications Manager does not support distribution of fixed-device (hardware) audio sources across Music On Hold servers within a media resource group.
Unacceptable Audio Quality with G.729a codec	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.

Restriction	Description
Cisco Unified Communications Manager System Support	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.
Multicast Support	The administrator can designate a Music On Hold server as either unicast or multicast, provided that resources exist to support multicast.
Caller-specific MOH Support	Caller-specific MOH is not supported when calls are received or transferred over QSIG tunneling-enabled SIP trunks.
MP3 Format Support	The Music On Hold feature does not support the MP3 format.
Interoperability between H.323 and SIP Protocols	Multicast MOH does not support interoperability between H.323 and SIP protocols.
SRTP Support	Multicast MoH audio streams are not encrypted and do not support SRTP.
Multicast Streams	MTPs do not support multicast streams.
Encryption of Multicast Music On Hold RTP Streams	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.
Fixed Music On Hold Device	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.
MOH Server Failure	Cisco Unified Communications Manager takes no action when a Music On Hold server fails during an active Music On Hold session.
Multicast MOH	When an MTP resource gets invoked in a call leg at a site that is using multicast MOH, Cisco Unified Communications Manager falls back to unicast MOH instead of multicast MOH.
Provisioning	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.
Header Values	• 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). • If both values are zero, or the only value is zero, the header in the incoming INVITE is ignored.

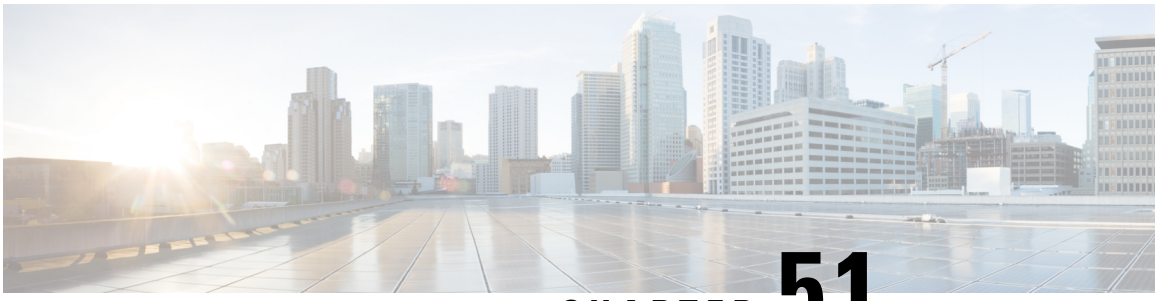
Restriction	Description
MOH Audio Source Identifier	• If you provide only one MOH audio source identifier in the SIP header, including if a comma appears before or after the MOH audio source identifier value, the same MOH ID is used for both user and network MOH. The SIP trunk populates both the user and the network MOH audio source identifiers in the SIP header so that Call Control always receive both values. • If there are more than two MOH audio source identifier values separated by a comma in the header, then the first two values are used. Subsequent values are ignored.
Administrators for Consistent Caller-specific MOH Configurations	Administrators are responsible to maintain consistent caller-specific MOH configurations when multiple Cisco Unified Communications Manager clusters are involved.
Original Incoming Caller	The original incoming caller to the call center cannot change during the course of the entire call.
MOH Information	The Music On Hold information is shared only across SIP trunks.

Music On Hold Troubleshooting

Music On Hold Does Not Play on Phone

Phone user cannot hear Music On Hold.

- G.729a codec is used with MOH for music, which may not provide acceptable audio quality.
- An MTP resource is invoked in a call leg at a site that is using multicast MoH.
- When an MTP resource gets invoked in a call leg at a site that is using multicast MoH, the caller receives silence instead of Music On Hold. To avoid this scenario, configure unicast MoH or Tone on Hold instead of multicast MoH.



CHAPTER 51

Self Care Portal

- Self Care Portal Overview, on page 637
- Self Care Portal Task Flow, on page 637
- Self Care Portal Interactions and Restrictions, on page 639

Self Care Portal Overview

From the Cisco Unified Communications Self Care Portal, users can customize features and settings for their phones. As the administrator, you control access to the portal. Before an end user can access the portal, you must add the user to the default **Standard CCM End Users** access control group, or to any access control group that has the **Standard CCM End Users** role assignment. In addition, users require their user ID, password, and the URL with which to access the portal. Users can access the portal via the following URL:

`http(s)://<server_name>:<port_number>/ucmuser/`

where:

- **`<server_name>`** represents the Unified Communications Manager IP address, hostname or fully qualified domain name
- **`<port_number>`** represents the port on which to connect. The port is optional, but is useful in firewall situations.
- **`ucmuser`** is a mandatory subpath that points to Self Care

Optionally, you can also configure enterprise parameters within Cisco Unified Communications Manager in order to assign which phone settings are available for end users to configure. For example, the **Show Call Forwarding** enterprise parameter determines whether users can configure Call Forward via the portal.

Self Care Portal Task Flow

Procedure

	Command or Action	Purpose
Step 1	Grant User Access to the Self Care Portal, on page 638	To access the portal, end users must be assigned to the **Standard CCM End Users** access

	Command or Action	Purpose
		control group or to any group that has the **Standard CCM End Users** role assignment.
Step 2	Configure the Self Care Portal Options, on page 638	Configure enterprise parameters in order to control what configuration options are available to users whom access the portal.

Grant User Access to the Self Care Portal

To access the portal, end users must be assigned to the **Standard CCM End Users** access control group or to any group that has the **Standard CCM End Users** role assignment.

Procedure

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

Step 2 Search for the user for whom you want to provide Self-Care access.

Step 3 In the **End User** section, ensure that the user has a password and PIN configured.

Usually these credentials are entered when a new user is added.

Step 4 In the **Permission Information** section, click **Add to Access Control Group**.

Step 5 Click **Find** and select the **Standard CCM End Users** group or a customized group that contains the **Standard CCM End Users** role.

Note
For information on how to edit access control groups, and role assignments for access control groups, refer to the "Manage User Access" chapter of the *Administration Guide for Cisco Unified Communications Manager*.

Step 6 Select **Save**.

Configure the Self Care Portal Options

Use this procedure to configure Self Care Portal enterprise parameters in order to control what configuration options are available to users whom access the portal.

Before you begin

Grant User Access to the Self Care Portal, on page 638

Procedure

Step 1 From Cisco Unified Communications Manager Administration, select **System** > **Enterprise Parameters**.

Step 2 Under **Self Care Portal Parameters**, set the **Self Care Portal Default Server** by selecting one of the available servers from the drop-down list.

This parameter determines which Cisco Unified CM server Jabber uses to display embedded Self Care options pages. If you select **None**, Jabber defaults to the Publisher.

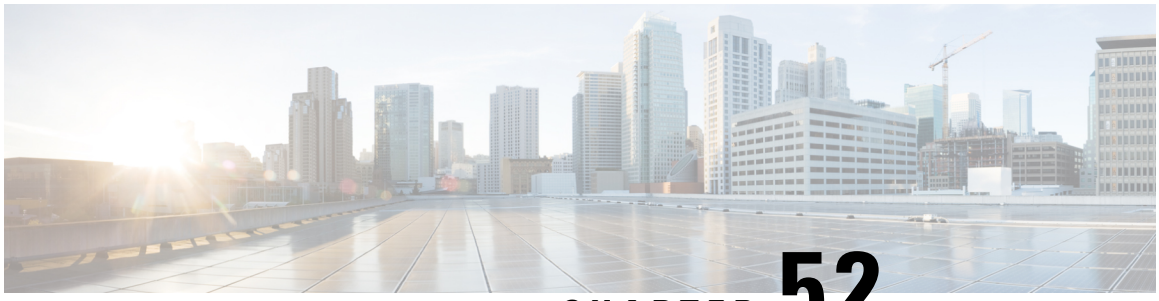
Step 3 Configure any of the remaining Self Care Portal Parameters to enable or disable features for the portal. For help with the fields, refer to the enterprise parameters help.

Step 4 Select **Save**.

Self Care Portal Interactions and Restrictions

The following table highlights feature interactions and restrictions with the Self-Care Portal.

Feature	Interaction or Restriction
Device Onboarding via Activation Codes	If you want users to be able to activate their phones via the Self-Care Portal, the **Show Phones Ready to Activate** enterprise parameter must be set to **True** (this is the default setting). With this feature, users can obtain their activation code by logging in to the Self-Care portal. They can either use the phone's video camera to scan the barcode, or they can enter the code manually on the phone in order to activate and register the phone. For more information on activation codes, see the "Device Onboarding via Activation Codes" chapter of the *System Configuration Guide for Cisco Unified Communications Manager*.
Authenticated user https request	When an authenticated user makes a request to `https://{CUCM_address}/ucmuser/hostAlive/{host}`, the following happens: • If the request is successful at getting `http:{host}/` or if the request can ping `{host}` Cisco Unified Communications Manager returns the string, "`true`". • If the request is unsuccessful, Cisco Unified Communications Manager returns the string "`false`".
Maximum Login for Extension Mobility	For end users to be able to configure this setting within the Self-Care Portal, an administrator must have checked the **Allow End User to set their Extension Mobility maximum login time** option in the associated User Profile of Cisco Unified CM Administration. If this option is selected within the User Profile, for all users whom use the profile, the Self-Care Portal setting overrides the administrator-configured values of the **Intra-cluster Maximum Login Time** and **Inter-cluster and Maximum Login Time** service parameters in Cisco Unified Communications Manager.



CHAPTER 52

Emergency Call Handler

- Emergency Call Handler Overview, on page 641
- Emergency Call Handler Prerequisites, on page 642
- Emergency Call Handler Task Flow, on page 642
- Interactions, on page 649
- Emergency Call Handler Troubleshooting, on page 651

Emergency Call Handler Overview

Emergency Call Handler helps you to manage emergency calls in your telephony network while following local ordinances and regulations.

When an emergency call is made the following is required:

- The emergency call must be routed to the local Public-Safety Answering Point (PSAP) based on the location of the caller.
- The caller's location information must be displayed at the emergency operator terminal. The location information can be obtained from an Automatic Location Information (ALI) database.

The caller's location is determined by the Emergency Location Identification Number (ELIN). An ELIN is a Direct Inward Dial (DID) number that the PSAP can dial to reconnect to the emergency caller if the emergency call is cut off or if the PSAP needs to talk to the caller again. The emergency call is routed to the PSAP based on the location information that is associated with this number.

For multiline phone systems, such as an office system, the ELIN can be associated with more than one telephone by grouping the phones in an ELIN group. An ELIN group in Emergency Call Handler identifies a location. The ELINs under this ELIN group must be mapped to the location in the ALI database.

Each location should have as many ELINs created as needed to support simultaneous emergency calls. For example, to support five simultaneous calls, five ELINs would be needed in an ELIN group.

Note Emergency Call Handler supports a maximum of 100 ELINs per cluster.

Make sure that the mapping of ELIN to the original called party is active until the ELIN is used for the next emergency call from the same location. If the ELIN mapping is not used, the DN will be active for a maximum period of 3 hours only.

The following types of phone are supported to use ELIN groups:

- SIP and SCCP IP phones
- CTI ports
- MGCP and SCCP analog phones
- H.323 phones

Emergency Call Handler Prerequisites

Example

Before deploying Emergency Call Handler in your network, we recommend that you test the ALI submission process. With your service provider's help, test that the PSAP can successfully call back into your network using the ALI data.

Reserve the ELIN number from your local PSAP. Ordinances and regulations can differ across different locations and across different companies, so research your security and legal needs before deploying this feature.

Emergency Call Handler Task Flow

Before you begin

- Review Emergency Call Handler Prerequisites, on page 642

Procedure

	Command or Action	Purpose
Step 1	Enable Emergency Call Handler, on page 643	Enable the Emergency Call Handler feature on Cisco Unified Communications Manager. Emergency Call Handler provides essential emergency call features and supports a limited number of locations with phone location assignment by static configuration. If you require advanced emergency call features, such as a greater amount of specific locations or dynamic location assignment, consider Cisco Emergency Responder.
Step 2	Configure Emergency Location Groups, on page 644	Configure an Emergency Location (ELIN) Group for a particular site or location.
Step 3	Add a Device Pool to an Emergency Location Group, on page 645	Configure device pools to use an Emergency Location (ELIN) Group.

	Command or Action	Purpose
Step 4	(Optional) Add Device to an Emergency Location Group, on page 645	Configure a particular device to use a particular Emergency Location (ELIN) Group. If you want to use the device pool ELIN Group that is associated for this device, you can ignore this section. **Note** Configurations that are made at the device level will overwrite any configurations that were made at the device pool level.
Step 5	Enable Route Patterns and Translation Patterns, on page 646	Enable the Emergency Location (ELIN) service for a route pattern or a translation pattern. **Caution** No Calling Party Transformation masks are set at the Gateway or Trunk, because these may transform the ELIN that is set by Emergency Call Handler. **Note** It is mandatory that you enable either route patterns or translation patterns, but it is possible to enable both.
Step 6	(Optional) Use the following procedures to perform bulk administration tasks on ELIN group information and phones: • Import Emergency Location Group Information, on page 647 • Export Emergency Location Group Information, on page 647 • Update Phones with a new Emergency Location Group, on page 648	This section provides information about the Bulk Administration tasks you can use to update ELIN group information and to add phones to new ELIN groups. For Bulk Administration, see the *Cisco Unified Communications Manager Bulk Administration Guide, Release 11.0(1)*.

Enable Emergency Call Handler

Enable the Emergency Call Handler feature on Cisco Unified Communications Manager. Emergency Call Handler provides essential emergency call features and supports a limited number of locations with phone location assignment by static configuration. If you require advanced emergency call features, such as a greater amount of specific locations or dynamic location assignment, consider Cisco Emergency Responder.

Note Do not enable this feature if you are already using an external emergency calling solution such as Cisco Emergency Responder.

If you decide to enable this feature, make sure you disable the external one.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Emergency Call Handler** > **Emergency Location Configuration**.

Step 2 From the Emergency Location Configuration window:

- To enable the Emergency Call Handler feature, check the **Enable Emergency Location (ELIN) Support** check box. The setting default is Disabled. When enabled, the settings related to this feature appear in the Related Settings pane. You must configure these settings for the feature to work. Refer to the tasks below for further details on how to configure these related settings.
- To disable the Emergency Call Handler feature, uncheck the **Enable Emergency Location (ELIN) Support** check box.

  Note
  If you disable this feature, all related settings that are configured will be removed. See the Related Settings Pane for all configured settings.

  Note
  If you want to disable the feature and you have more than 500 devices associated with ELIN Groups, then you must manually delete the associations until there are fewer than 500 associations before you can disable the feature.

Step 3 Click **Save**.

Configure Emergency Location Groups

Configure an Emergency Location (ELIN) Group for a particular site or location.

Before you begin

Enable Emergency Call Handler, on page 643

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Emergency Call Handler** > **Emergency Location (ELIN) Group**.

Step 2 In the **Emergency Location (ELIN) Group Configuration** window, enter a name for the group in the **Name** field.

Step 3 In the **Number** field, enter the pool of DID numbers that are registered in the Public Safety Answering Point (PSAP).

Step 4 Click **Save**.

Add a Device Pool to an Emergency Location Group

Configure device pools to use an Emergency Location (ELIN) Group.

Before you begin

Configure Emergency Location Groups, on page 644

Procedure

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

Step 2 In the **Find and List Device Pools** window, if you are adding an existing device pool, click **Find** and choose the device pool from the list. If you are adding a new device pool click **Add New**.

Step 3 In the **Device Pool Configuration** window, choose the ELIN group to which you want to add the device pool from the **Emergency Location (ELIN) Group** drop-down list. If you are adding a new device pool, fill out any other required fields.

Step 4 Click **Save**.

Add Device to an Emergency Location Group

Configure a particular device to use a particular Emergency Location (ELIN) Group. If you want to use the device pool ELIN Group that is associated for this device, you can ignore this section.

Note Configurations that are made at the device level will overwrite any configurations that were made at the device pool level.

Note The devices that you add to the ELIN Group, should be added to the ELIN Group that represents the particular location at which those devices are located.

Before you begin

Add a Device Pool to an Emergency Location Group, on page 645

Procedure

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

Note
If you are using a type of phone that is not an IP phone, go to the relevant configuration page for that type of phone.

Step 2 In the **Find and List Phones** window, if you are adding an existing device, click **Find** and choose the device you want to configure from the list. If you are adding a new device, click **Add New**.

Step 3 If you are adding a new phone, choose the type of phone you want to add from the **Phone Type** drop-down list and click **Next**.

Step 4 In the **Phone Configuration** window, choose the ELIN group to which you want to add the device from the **Emergency Location (ELIN) Group** drop-down list. If you are adding a new device, fill out any other required fields.

Step 5 Click **Save**.

Enable Route Patterns and Translation Patterns

Enable the Emergency Location (ELIN) service for a route pattern or a translation pattern.

Note It is mandatory that you enable either route patterns or translation patterns, but it is possible to enable both.

Before you begin

Add Device to an Emergency Location Group, on page 645

Procedure

Step 1 From Cisco Unified CM Administration, choose one of the following:

- To enable a route pattern, choose **Call Routing** > **Route/Hunt** > **Route Pattern**.
- To enable a translation pattern, choose **Call Routing** > **Translation Pattern**.

Step 2 In the **Find and List Route Patterns** or **Find and List Translation Patterns** window, click **Find** and choose a route pattern or translation pattern from the list.

Step 3 In the **Route Pattern Configuration** or **Translation Pattern** Configuration window, check the **Is an Emergency Services Number** check box.

Note
Check this check box only if you are using Emergency Call Handler and not another external emergency calling solution such as Cisco Emergency Responder.

Step 4 Click **Save**.

Bulk Administration of Emergency Location Groups and Phones

- Bulk Administration of Emergency Location Groups and Phones Task Flow, on page 647

Bulk Administration of Emergency Location Groups and Phones Task Flow

This section provides information about the Bulk Administration tasks you can use to update ELIN group information and to add phones to new ELIN groups. For more information about Bulk Administration, see the *Cisco Unified Communications Manager Bulk Administration Guide, Release 11.0(1)*.

Note Before you perform these procedures, make sure that you have enable the Emergency Call Handler feature. See Enable Emergency Call Handler, on page 643.

Procedure

	Command or Action	Purpose
Step 1	Import Emergency Location Group Information, on page 647	Import Emergency Location (ELIN) Group information using the Bulk Administration Tool.
Step 2	Export Emergency Location Group Information, on page 647	Export Emergency Location (ELIN) Group information using the Bulk Administration Tool.
Step 3	Update Phones with a new Emergency Location Group, on page 648	Find and list multiple phones and configure them with a new Emergency Location (ELIN) Group.

Import Emergency Location Group Information

Import Emergency Location (ELIN) Group information using the Bulk Administration Tool.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Bulk Administration** > **Import/Export** > **Import**.

Step 2 From the **File Name** drop-down list, choose the name of the .tar file you want to import, and click **Next**.

Step 3 The **Import Configuration** section lists all the components of the .tar file. Check the ELIN Group-related check boxes for the options that you want to import.

Step 4 Choose to run the job immediately or later by clicking the corresponding radio button.

Step 5 To create a job for importing the selected data, click **Submit**. A message in the Status section notifies you know that the job was submitted successfully.

Step 6 Use the Job Scheduler option in the Bulk Administration main menu to schedule or activate this job.

Export Emergency Location Group Information

Export Emergency Location (ELIN) Group information using the Bulk Administration Tool.

Before you begin

Import Emergency Location Group Information, on page 647

Procedure

Step 1 From Cisco Unified CM Administration, choose **Bulk Administration** > **Import/Export** > **Export**.

Step 2 In the **Export Data** window, in the **Job Information** pane, enter the .tar file name, without the extension, in the **Tar File Name** field. BPS uses this filename to export the configuration details.

Note
All files that are exported at the same time get bundled together (.tar) and can be downloaded from the server.

Step 3 To export ELIN Group information, check the **Elin Group** check box on the **Select items to Export** pane.

Step 4 (Optional) Perform these steps:

- To export device pools with ELIN Groups configured, check the **Device Pools** check box.
- To export phones with ELIN Groups configured, check the **Phone** check box.

Step 5 In the **Job Descripton** field, enter the description that you want to override for the job. Export Configuration is the default description.

Step 6 You can choose to run the job immediately or later by clicking the corresponding radio button.

Step 7 To create a job for exporting the selected data, click **Submit**. A message in the **Status** pane notifies you that the job was submitted successfully.

Step 8 Use the Job Scheduler option in the Bulk Administration main menu to schedule or activate this job.

Update Phones with a new Emergency Location Group

Find and list multiple phones and configure them with a new Emergency Location (ELIN) Group.

Before you begin

Export Emergency Location Group Information, on page 647

Procedure

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

Step 2 In the **Find and List Phones To Update** window, set the parameters for your search and click **Find**.

Note
To update all phones, click **Find** and do not specify a query.

Step 3 The **Find and List Phones To Update** window displays the details of the phones that you chose. Click **Next**.

Step 4 In the **Update Phones** window, check the **Emergency Location (ELIN) Group** check box, and choose a new ELIN Group from the drop-down list.

Step 5 Click **Submit**.

Interactions

Feature	Interaction
Do Not Disturb Call Reject	Calls made by PSAP CallBack will overwrite a Do Not Disturb (DND) configuration of a destination device. If DND Call Reject is enabled, when the emergency number is dialed using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected and the ELIN is called back using PSAP CallBack, the call is routed to the phone irrespective of the phone's DND settings.
Call Forward All	Calls made by PSAP CallBack will overwrite Call Forward All (CFA) settings of the destination device. If a phone has CFA enabled and if the emergency number using the translation pattern is dialed, an ELIN will be associated for this outbound emergency call. If the call is disconnected and the ELIN is called back using PSAP CallBack, the call is routed to the phone irrespective of the phone's CFA settings.
Single Number Reach	PSAP CallBack will ignore the Single Number Reach (SNR) configuration. When a phone has SNR enabled with the Remote Destination pointing to a mobile number. If the emergency number is dialed using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected, and the ELIN number is called back using PSAP CallBack, the call is routed to the phone and not to the remote destination.

Feature	Interaction
Extension Mobility	PSAP CallBack call will consider Extension Mobility (EM) status. If you log in with EM profile credentials and dial the emergency number using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected and the ELIN where the user is still logged in is called back using PSAP CallBack, the call is routed to the device which initiated the call. **Note** This is the device on which the user is still logged in.
	PSAP CallBack will fail if a user logs out of EM before a PSAP CallBack is performed. When a user logs in with EM profile credentials, and the emergency number is dialed using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected and is called back using PSAP CallBack, if the user has since logged out, the call will not route to the device that initiated the call and will fail.
	PSAP CallBack with a user logged in on a different device. When a user logs in with EM profile credentials at Phone A and dials the emergency number using the translation pattern, an ELIN will be associated for this outbound emergency call. If the call is disconnected, the user should log out from Phone A. If the user then logs in to another phone, Phone B, with the same profile, and the ELIN is called back using PSAP CallBack, the call is then be routed to Phone B with normal priority, meaning CFA settings will be ignored and DND settings will not be ignored.
	PSAP CallBack call with multiple logins. When a user logs in with EM profile credentials at Phone A and dials the emergency number using the translation pattern, an ELIN number will be associated for this outbound emergency call. If the call is disconnected and the user logs in to another phone, Phone B, with the same profile while the user is still logged in on Phone A, and the ELIN is called back using PSAP CallBack, then the call is routed to Phone A only, the device on which the call originated.
Device Mobility	A roaming device will use the Roaming Device Pool's ELIN Group for an outbound emergency call. Move a device with Device Mobility enabled from its home location to the Roaming location, a change in IP subnet, so that it gets associated with the Roaming device pool. If the emergency number is dialed using the translation pattern, an ELIN is associated for this outbound emergency call. The ELIN belongs to the ELIN Group that is associated with the Roaming Device Pool.

Feature	Interaction
Shared Lines	PSAP CallBack rings only on the device which made the emergency call even if the line is shared by different devices. Phone A and Phone B share a Directory Number (DN). If the emergency number is dialed using the translation pattern, an ELIN is associated for this outbound emergency call. If the call is disconnected, and the ELIN is called back using PSAP CallBack, the call is routed to Phone A only, the device from which the call originated.

Emergency Call Handler Troubleshooting

About Emergency Call Handler Troubleshooting Scenarios

This section provides information about some Emergency Call Handler troubleshooting scenarios in the following areas:

- Configuration Scenarios
- Outgoing Calls Scenarios
- Incoming Calls Scenarios

Configuration Scenarios

Emergency Calls Get Busy Signals and Are Not Routed

Problem:

Emergency calls get busy signals and are not routed.

Solution:

If a user who is dialing the emergency call is running a reorder tone, perform the following checks:

- Check whether the translation or route pattern for the emergency call has been used. This may require checking for the device or phone on CSS.
- Check whether the **Is an Emergency Services Number** check box has been checked for the translation or route pattern of the emergency call, and that it is correctly routing to the gateway.

If the user who is dialing the emergency call is not reaching the correct gateway or Public Service Answering Point (PSAP), check that the settings or device pool settings for the phone or device are configured with the correct Emergency Location (ELIN) Group.

Emergency Location Numbers Are Dialed from Outside Running a Reorder Tone

Problem:

Emergency Location (ELIN) numbers are dialed from outside while running a reorder tone.

Cause:

In this case the ELINs have been set as DID which is used to identify a caller's location. This should not be used on any phone or for any other purpose.

Solution:

Check the ELIN configuration information, and unset the ELINs that have been set as DID.

Outgoing Calls Scenarios

Outgoing Emergency Call Does Not Contain Calling Party as Emergency Location Number

Problem:

An outgoing emergency call does not contain the calling party as an Emergency Location (ELIN) number.

Cause:

The translation pattern or route pattern for this ELIN was not configured correctly.

Solution:

Check that the translation pattern or route pattern settings are correctly configured for this ELIN, and make sure that the **Is an Emergency Services number** check box is checked on the relevant translation pattern or route pattern configuration page.

Outgoing Emergency Call Contains Modified Emergency Location Number

Problem:

An outgoing emergency call contains a modified Emergency Location (ELIN) number.

Cause:

The outgoing trunk or route list contains extra transformations that are not required for ELINs.

Solution:

Check the transformations that were applied for the call, and make sure that only the required transformations for ELINs are present on the outgoing trunk or route list.

Incoming Calls Scenarios

Incoming PSAP Callback Call Fails

Problem:

An incoming PSAP Callback call fails.

Cause:

The device that made the original emergency call was not registered correctly.

Solution:

Check whether the device that made the original emergency call is still registered and whether any Extension Mobility is involved.

Incoming PSAP CallBack Call is Not Routed as Expected

Problem:

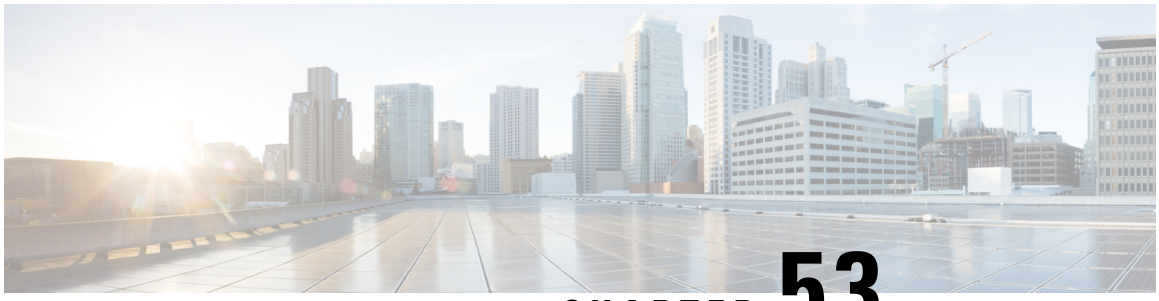
An incoming PSAP CallBack call does not get routed as expected.

Cause:

The Emergency Location (ELIN) number does not match the number of the original dialed party.

Solution:

For an ELIN to be successfully reverse mapped to the original dialed party, these two numbers must match. If there are already transformations at the incoming Gateway or Trunk and significant digits configured, make sure that the final transformed called party matches the ELIN number.



CHAPTER 53

Emergency Call Handling with RedSky

- Emergency Call Handling with RedSky Overview, on page 655
- Emergency Call Handling Configuration Task Flow, on page 656

Emergency Call Handling with RedSky Overview

Important This feature is applicable from Releases 12.5(1)SU6 and 14SU2 onwards.

The RedSky solutions integrated with Unified Communications Manager allow the clients to have an active location for 9-1-1 emergency calling coverage for their entire workforce, whether on campus or remote and send the calls to emergency responders.

The endpoints store the location URI received from the RedSky server as a response to HTTP Enabled Location Delivery (HELD) request. When an emergency number 9-1-1 is dialed from Webex, the Unified Communications Manager obtains the previously saved location URI as Geolocation header in INVITE message and routes the calls to the RedSky server with outgoing INVITE containing the location URI as Geolocation header corresponding to the location of the called device. RedSky server replaces with the right ELIN and sends the call to any Public Safety Answering Points (PSAP) for an emergency transmit. E911 Anywhere simultaneously sends call notifications including SMS text, Email, and Security desk screen alerts.

The Cisco Emergency Responder automatically finds and tracks the dispatchable locations of all your devices as they move throughout the enterprise so you can comply with E911 regulations. Emergency Responder tracks Cisco IP Phones through Switch Port or Access Point or IP Subnet or Manually configured. Emergency Responder maintains the status of the phones (On-premises, Off-premises, unlocated), and passes on any Automatic Location Information (ALI) or ELIN information to RedSky. Phone users rely on Unified CM to route their emergency calls to RedSky and the designated emergency provider.

For Off-premises phones, if the user's phone's current location has not been previously defined, the user is directed to the Emergency Responder Off-Premises User web page to create a new location. After the new location has been defined and the address has been confirmed, emergency calls placed from off-premises phones will then be completed through the RedSky.

Note We recommend that when an employee is working on-premises at an organization site, the user’s location should be defined by the calling system administrator.

Emergency Call Handling Configuration Task Flow

The administrator can use the following task to have a dynamic location for 9-1-1 emergency calling and transfer the call to emergency responders.

Procedure

	Command or Action	Purpose
Step 1	Configure RedSky Server	Create a SIP Trunk for routing the call to the RedSky server.
Step 2	Configure Service Profile	Add the Service Profile details of an end-user for emergency calls.
Step 3	Assign the Service Profile	Assign the created service profile to the Webex client end-user.
Step 4	Setting Up the SIP Route Pattern for Routing Calls	Create SIP Route Pattern with the domain name and associate the same with the previously created SIP trunk.

Configure RedSky Server

Use this procedure to create a SIP Trunk for routing the call to the RedSky server.

Note Steps 7, 8 and 9 are only needed during on-premises integration.

Procedure

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

Step 2 Click **Add New**.

Step 3 From the **Trunk Type** drop-down list, choose **SIP Trunk**.

Step 4 From the **Protocol Type** drop-down list, choose the type of SIP trunk that matches your deployment and click **Next**.

Step 5 In the **SIP Information** area, enter an IPv4 address of the RedSky server, fully qualified domain name, or DNS SRV record for the server or endpoint that you want to connect to the SIP trunk in the **Destination Address** text box.

Step 6 From the **SIP Trunk Security Profile** drop-down, assign a security profile. If you don't select this option, a SIP Trunk Security Profile nonsecure profile will be assigned.

Step 7 (Optional) From the **SIP Profile** drop-down list, assign a Ping option enabled RedSky SIP profile.

Step 8 (Optional) In the **Normalization Script** area, from the **Normalization Script** drop-down, choose **redsky-alternate-id-interop**.

Step 9 (Optional) For **Parameter Name** and **Parameter Value**, enter the respective information.

The following inputs are supported for **Parameter Name**:

- **RedSky-CustomerID**—This is a mandatory field. It is the HELD ID from RedSky admin page. This is used to identify the customer account for the calling party.
- **Alternate-Callback-Number**—This is an optional field. This field inserts an optional callback number for emergency calls. It should be used for callers that do not have Direct Inward Dial (DID) numbers for callback.
- **Ext-Length**—This is an optional field. This parameter is used for customers with non-E.164 numbering convention. The parameter will enters the non-E.164 into the RedSky E911-User-ID header.
- **Agent-Ext**—This is an optional field. This parameter identifies agent extensions based on the matching digits. Populating this parameter puts the agent calling party into the RedSky E911-User-ID header.

  The script does not only look for leading digits in the extension it looks at. For example, if Agent-Ext is set to "5", 12345678 will match, although 12345678 does not have a 5 as a leading digit.

  If Agent-Ext is set to 100200, then 123410020088 will match although 100200 is not a leading digit.

  If Agent-Ext is set to 12, then 446658787 will not match because it does not contain 12.

Step 10 Click **Save**.

Configure Service Profile

Use this procedure to add the Service Profile details of an end-user for emergency calls.

Before you begin

- You must create a SIP Trunk with the destination as the RedSky server and a SIP Profile with Ping options enabled. A SIP route pattern must be created with the required domain name (RedSky), and it is associated with the trunk created previously.
- A service profile is applied for a given device only when the owner's user ID is specified.



Note The Emergency Call Handling with RedSky feature can be done without using Cisco Emergency Responder emergency call routing.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **User Settings** > **Service Profile**.

Step 2 Click **Add New**.

Step 3 Enter a **Name** and **Description** for the chosen Service Profile Configuration.

Note

For each UC service that you want to be a part of this profile, assign the **Primary**, **Secondary**, and **Tertiary** connections for that service. The fields in the **Service Profile Configuration** window vary depending on which UC service you configure.

Step 4 In the **Emergency Calling Profile** section, perform the following:

a) Check Enable Emergency Calling to enable configuration parameters to endpoints and soft clients to update location and send emergency calls to the Emergency Calling Service Provider.
b) Enter the Company ID and Passphrase provided by the Emergency Calling Service Provider when the account is created, and service is enabled in the **Organization ID** and **Secret** field. For example, 32-character alphanumeric string provided by RedSky.
c) Enter the passphrase required by the Emergency Calling Service Provider Authorization Service in the **Secret** field. For example, 16-character alphanumeric string provided by RedSky.
d) Enter the URL that the device uses to request and set the location in the **Location URL** field.
e) Enter the **Emergency Service Numbers**. By default, 911, 933 is entered with a comma separating each number.

Note
When Webex client dials any emergency pattern configured in Emergency Numbers, it will be routed with Geolocation headers to the RedSky server configured in the SIP Trunk.

Step 5 Complete the remaining fields in the **Service Profile Configuration** window. For detailed field descriptions, see the online Help.

Step 6 Click **Save**.

Assign the Service Profile

Use this procedure to assign the created service profile to the Webex client end-user. If Webex is not registered to Unified CM, the end-user will not be active and does not route the emergency calls to RedSky.

You can apply the Service Profile to an end-user to assign the UC services configuration settings in the Service Profile to that end user. You can configure different service profiles for different groups of users in the organization so that each group of users has the right services configured for their job.

Procedure

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

Step 2 In the **Find and List Users** window, perform either of the following:

a) Click **Add New** to configure a new user.
b) Specify the filters in the **Find User Where** field and then click **Find** to retrieve a list of users.

Note
For more information on associating a device with a user, see the Associate Devices to End User section in Cisco Emergency Responder Administration Guide .

Step 3 In the **Service Settings** section, select the RedSky Service Profile from the **UC Service Profile** drop-down list.

Step 4 Complete the remaining fields in the **End User Configuration** window. For detailed field descriptions, see the online help.

Step 5 Click **Save**.

Setting Up the SIP Route Pattern for Routing Calls

Use this procedure to create SIP Route Pattern with the domain name and associate the same with the previously created SIP trunk.

All emergency calls that are routed to the Emergency Provider must match a route pattern. The route pattern directs the call to a Route Group, Route List, and SIP Trunk or PRI gateway that can reach the RedSky server.

PRI - RedSky provides the customer an account-specific access number. In this case, the number is the customer ID and the CALLING PARTY is the user reference. It follows traditional RP/RG/RL/GW redundancy. The calling party number must match the RedSky user's ID.

We recommend using SIP Trunks to connect with the RedSky server. For dedicated instances, this is the default method. For customers having an on-premises deployment of Unified Communications Manager on-premises, you must configure the SIP Trunk, Route Group and Route List before creating the route pattern that will be used to reach the RedSky server.

If using a SIP trunk, the administrator must use a predefined LUA script to ensure proper customer identification. For Unified CM deployments, you must upload the script and apply it to the SIP Trunk. The LUA script allows for only one parameter, which is the RedskyOrgID.

Procedure

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

Step 2 Click **Add New** to add the RedSky route pattern.

Step 3 From the **Pattern Usage** drop-down, choose **Domain Routing**.

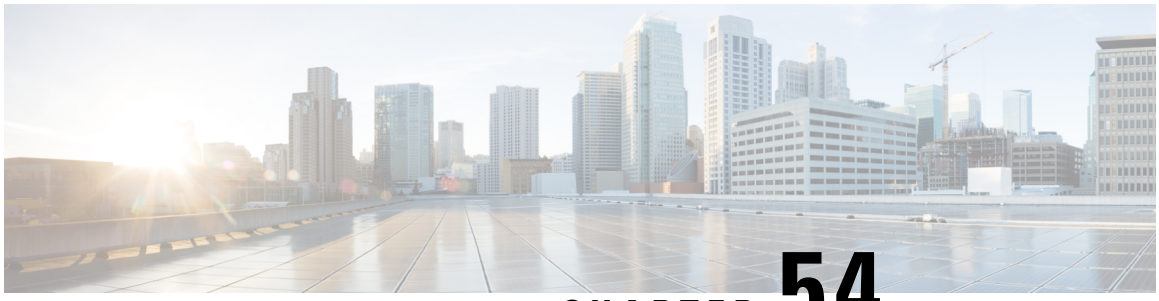
Step 4 Enter the route string in the **IPv4 Pattern** or **IPv6 Pattern** field depending on whether you are deploying the IPv4 or IPv6 address.

Step 5 Choose a RedSky SIP trunk in the **SIP Trunk/Route List*** drop-down.

Step 6 (Optional) Click the **Edit** link to view or change the Trunk Configuration details.

Step 7 Complete the remaining fields in the SIP Route Pattern Configuration window. For detailed field descriptions, see the online Help.

Step 8 Click **Save**.



CHAPTER 54

Enterprise Groups

- Enterprise Groups Overview, on page 661
- Enterprise Groups Prerequisites, on page 662
- Enterprise Groups Configuration Task Flow, on page 662
- Enterprise Groups Deployment Models (Active Directory), on page 666
- Enterprise Groups Limitations, on page 668

Enterprise Groups Overview

When Enterprise Groups is configured, Cisco Unified Communications Manager includes user groups when it synchronizes its database with an external LDAP directory. In Cisco Unified CM Administration, you can view synced groups in the User Groups window.

This feature also helps administrators to:

- Provision users with similar characteristics traits with a comment set of features (for example, the sales and accounting teams).
- Target messages to all users in a specific group.
- Configure uniform access for all members of a specific group

This feature also helps Cisco Jabber users to quickly build contact lists of users who shares common traits. Cisco Jabber users can search the external LDAP Directory for user groups and then add them to their contact list. For example, a Jabber user can search the external LDAP directory and add the sales group to a contact list, thereby adding all of the sales team members into the contact list as well. If the group gets updated in the external directory, the user's contact list is updated automatically.

Enterprise Groups is supported with Microsoft Active Directory on Windows as the external LDAP directory.

Note If you disable the Enterprise Groups feature, Cisco Jabber users cannot search for enterprise groups or see the groups that they already added to their contact lists. If a user is already logged in when you disable the feature, the group will be visible until the user logs out. When the user logs in again, the group will not be visible

Security Groups

Security Groups are a subfeature of Enterprise Groups. Cisco Jabber users can also search for, and add, security groups to their contact list. To set up this feature, administrators must configure a customized LDAP filter and apply it to the configured LDAP directory sync. Security Groups are supported with Microsoft Active Directory only.

Maximum Allowed Entries

When configuring Enterprise Groups, make sure that you configure contact list maximums that handle groups

- The maximum number of entries that are allowed in a contact list is the sum of the number of entries in the contact list and the number of entries in groups that are already added to the contact list.
- Maximum entries in contact list = (number of entries in contact list) + (number of entries in groups)
- When the Enterprise Groups feature is enabled, Cisco Jabber users can add the groups to the contact list if the number of entries in the contact list is less than the maximum allowed entries. If the maximum allowed entries is exceeded while the feature is disabled, the users are not restricted until the feature is enabled. If the user continues to be logged in after the feature is enabled, no error message is displayed. When the user logs out and logs in again, an error message is displayed that asks the users to clear the excess entries.

Enterprise Groups Prerequisites

This feature assumes that you already have an LDAP Directory sync schedule configured with the below conditions. For details on how to configure an LDAP Directory sync, see the "Import Users from LDAP Directory" chapter of the *System Configuration Guide for Cisco Unified Communications Manager*.

- The Cisco DirSync service must be activated
- The LDAP Directory sync must include both users and groups
- Regular LDAP Directory syncs, as configured with the **LDAP Directory Synchronization Schedule** must be scheduled.

Supported LDAP Directories

Only Microsoft Active Directory is supported with enterprise groups.

Enterprise Groups Configuration Task Flow

Complete these tasks to configure the Enterprise Groups feature.

Procedure

	Command or Action	Purpose
Step 1	Verify Group Sync from LDAP Directory, on page 663	Confirm that your LDAP Directory sync includes both users and groups.

	Command or Action	Purpose
Step 2	Enable Enterprise Groups, on page 663	Complete this task to enable Cisco Jabber users to search for enterprise groups in Microsoft Active Directory and add them to their contact lists.
Step 3	Enable Security Groups, on page 664	(Optional) If you want Cisco Jabber users to be able to search for and add security groups to their contact lists, complete this task flow.
Step 4	View User Groups, on page 666	(Optional) View enterprise groups and security groups that are synchronized with Cisco Unified Communications Manager database.

Verify Group Sync from LDAP Directory

Use this procedure to confirm that your LDAP Directory sync includes users and groups.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Server** > **LDAP** > **LDAP Directory**.

Step 2 Click **Find** and select the LDAP directory from which you are syncing enterprise groups.

Step 3 Confirm that the **Synchronize** field has **Users and Groups** selected.

Step 4 Complete any remaining fields in the LDAP Directory configuration window. For help with the fields and their settings, refer to the online help.

Step 5 Click **Save**.

Enable Enterprise Groups

Configure the system to include enterprise groups in LDAP Directory syncs.

Procedure

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

Step 2 Under **User Management Parameters**, set the **Directory Group Operations on Cisco IM and Presence** parameter to **Enabled**.

Step 3 Enter a value for the **Maximum Enterprise Group Sized to allow Presence Information** parameter. The permitted range is 1 to 200 users with a default value of 100 users.

Step 4 From the **Syncing Mode for Enterprise Groups** drop-down list configure the LDAP sync that you want to perform at regular intervals: **None**, **Differential Sync**, **Full Sync**.

Note
Refer to the enterprise parameter help for additional assistance in configuring these fields.

Step 5 Click **Save**.

Enable Security Groups

If you want to allow Cisco Jabber users to be able to add a security group to their contact list, complete these optional tasks to include security groups in an LDAP Directory sync.

Note Security group sync is supported from Microsoft Active Directory only.

Note You cannot add new configurations into an existing LDAP Directory configuration in Cisco Unified Communications Manager where the initial sync has already occurred.

Procedure

	Command or Action	Purpose
Step 1	Create Security Group Filter, on page 664	Create an LDAP filter that filters both directory groups and security groups.
Step 2	Synchronize Security Groups from LDAP Directory, on page 665	Add your new LDAP filter to an LDAP Directory sync.
Step 3	Configure Cisco Jabber for Security Groups, on page 665	Update existing service profiles to give Cisco Jabber users whom are associated to that service profile access to search and add security groups.

Create Security Group Filter

Create an LDAP filter that filters security groups.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **LDAP** > **LDAP Filter**.

Step 2 Click **Add New**.

Step 3 Enter a unique **Filter Name**. For example, `syncSecurityGroups`.

Step 4 Enter the following **Filter**: `(&(objectClass=group)(CN=*))`.

Step 5 Click **Save**.

Synchronize Security Groups from LDAP Directory

Add your Security Group filter to an LDAP Directory sync and complete a sync.

Note You cannot add new configurations into an existing LDAP Directory configuration in Cisco Unified Communications Manager if the initial LDAP sync has already occurred.

Note For detailed information on how to set up a new LDAP Directory sync, see the "Configure End Users" part of the *System Configuration Guide for Cisco Unified Communications Manager*.

Before you begin

Create Security Group Filter, on page 664

Procedure

Step 1 In Cisco Unified CM Administration, choose **System** > **LDAP** > **LDAP Directory**.

Step 2 Do one of the following:

- Click Add New to create a new LDAP Directory.
- Click **Find** and select the LDAP Directory from which the security groups will be synchronized.

Step 3 From the **LDAP Custom Filter for Groups** drop-down list, select the security group filter that you created.

Step 4 Click **Save**.

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

Step 6 Click **Perform Full Sync Now** to synchronize immediately. Otherwise, security groups will be synchronized when the next scheduled LDAP sync occurs.

Configure Cisco Jabber for Security Groups

Update existing service profiles to allow Cisco Jabber users whom are associated to that service profile to add security groups from an LDAP directory to their contact lists.

Note For information on how to set up new service profiles and assign them to Cisco Jabber users, see the "Configure Service Profiles" chapter of the *System Configuration Guide for Cisco Unified Communications Manager*.

Before you begin

Synchronize Security Groups from LDAP Directory, on page 665

Procedure

Step 1 Complete any remaining fields in the **Service Profile Configuration** window. For help with the fields and their settings, refer to the online help.

Step 2 Click **Find** and select the service profile that your Jabber users use.

Step 3 Under **Directory Profile**, check the **Allow Jabber to Search and Add Security Groups** check box.

Step 4 Click **Save**.
Cisco Jabber users who are associated to this service profile can now search and add security groups.

Step 5 Repeat this procedure for all service profiles that your Cisco Jabber users use.

View User Groups

You can view the enterprise groups and security groups that are synchronized with the Cisco Unified Communications Manager database using the following steps.

Procedure

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

Step 2 Enter search criteria and click **Find**.
A list of user groups that match the search criteria is displayed.

Step 3 To view a list of users that belong to a user group, click on the required user group.
The **User Group Configuration** window appears.

Step 4 Enter search criteria and click **Find**.
A list of users that match the search criteria is displayed.

If you click on a user in the list, the **End User Configuration** window appears.

Enterprise Groups Deployment Models (Active Directory)

The Enterprise Groups feature offers two deployment options for Active Directory.

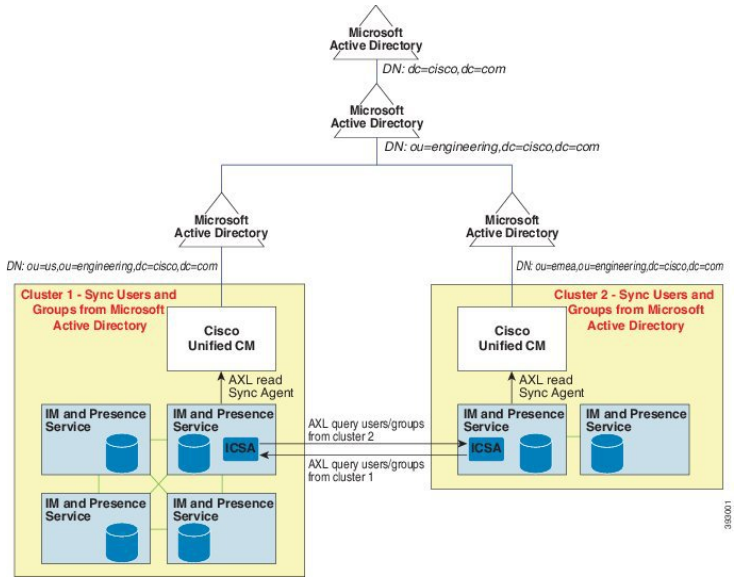


Important Ensure that Cluster 1 and Cluster 2 have a unique set of UserGroup, UserGroupMember, and UserGroupWatcherList records before synchronizing data through the Cisco Intercluster Sync Agent service. If both the clusters have unique sets of records, both the clusters will have a super set of all the records after synchronization.

Enterprise Groups Deployment Model 1

In this deployment model, Cluster 1 and Cluster 2 synchronize different subsets of users and groups from Microsoft Active Directory. The Cisco Intercluster Sync Agent service replicates the data from Cluster 2 into Cluster 1 to build the complete database of users and groups.

Figure 12: Enterprise Groups Deployment Model 1



Enterprise Groups Deployment Model 2

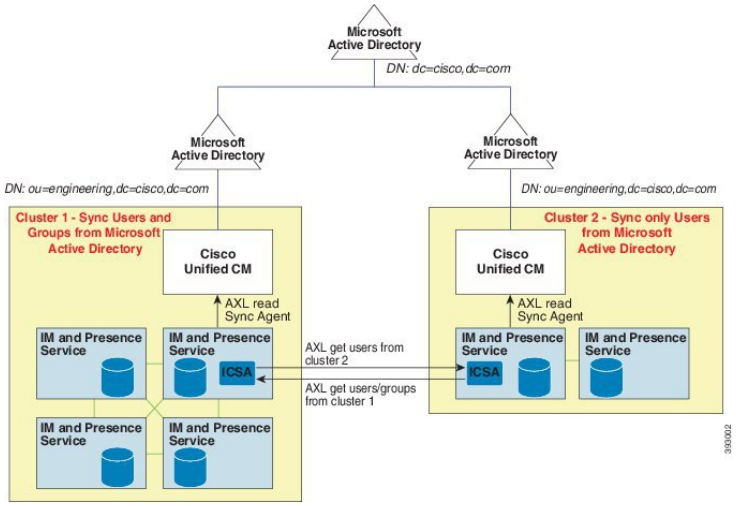
In this deployment model, Cluster 1 synchronizes all the users and groups from Microsoft Active Directory. Cluster 2 synchronizes only users from Microsoft Active Directory. The Cisco Intercluster Sync Agent service replicates groups information from Cluster 1 into Cluster 2.

Caution If you are using this deployment model, ensure that you synchronize the groups data in only one cluster. The Enterprise Groups feature will not work as expected if you fail to do so.

You can verify your configuration on the **Cisco Unified CM IM and Presence Administration** > **Presence** > **Inter-Clustering** window.

Check the status of the **Enterprise Groups LDAP Configuration** parameter in the Inter-cluster peer table. **No conflict found** means there are no misconfigurations between peers. If there are conflicts found, click the Enterprise GroupConflicts link, and click the **details** button which appears. This opens a Reporting window for a detailed report.

Figure 13: Enterprise Groups Deployment Model 2



Enterprise Groups Limitations

Table 60: Enterprise Groups Limitations

Limitation	Description
Block Everyone	When a Cisco Jabber user enables the "Block Everyone" feature from within their Cisco Jabber policy settings, the block prevents other Jabber users from viewing or exchanging IMs and Presence with the blocking user, unless they are listed as a contact in the blocking user's contact list. For example, a Cisco Jabber user (Andy) has enabled Block everyone within his personal Jabber settings. The following list breaks down how Andy's block affects other Jabber users whom may or may not be included in Andy's personal contact list. In addition to the block, Andy has a personal contact list that: • Includes Bob—Because Bob is in Andy's personal contact list, he can still send IMs and view Andy's presence despite the block. • Omits Carol—Carol cannot view Andy's presence or send IMs due to the block.. • Omits Deborah as a personal contact. However, Deborah is a member of an enterprise group that Andy has listed as a contact—Deborah is blocked from viewing Andy's presence or sending IMs to Andy. Note that Deborah is blocked from viewing Andy's presence, or sending IMs to Andy, despite the fact that she is a member of an enterprise group in Andy's contact list. For additional details on enterprise group contacts behavior, see CSCvg48001.

Limitation	Description
Intercluster peering with a 10.x cluster	Enterprise Groups is supported for releases 11.0(1) and higher. If the synced group includes group members from a 10.x intercluster peer, users on the higher cluster cannot view the presence of synced members from the 10.x cluster. This is due to database updates that were introduced in 11.0(1) for the Enterprise Groups sync. These updates are not a part of the 10.x releases. To guarantee that users homed on the higher cluster can view the presence of group members homed on the 10.x cluster, users on the higher cluster should manually add the 10.x users to their contact lists. There are no presence issues for manually added users.
Multilevel grouping	Multilevel grouping is not allowed for the group sync.
Group-only synchronization	When a user group and users are present in the same search base, group-only synchronization is not allowed. Instead, the user group as well as the users are synchronized.
Maximum number of user groups	You can synchronize a maximum of 15000 user groups from Microsoft Active Directory server to the Unified Communications Manager database. Each user group can contain from 1 to 200 users. You can configure the exact amount on the **Cisco Unified CM IM and Presence Administration** > **System** > **Service Parameters** window. The maximum number of user accounts in the database cannot exceed 160,000.
User group migration	If a user group is moved from one organization unit to another, you must perform a full sync on the original unit followed by a full sync on the new unit.
Local groups	Local groups are not supported. Only groups synchronized from Microsoft Active Directory are supported.
Group members not assigned to IM and Presence Service nodes	Group members that are not assigned to IM and Presence Service nodes display in the contact list with the presence bubble greyed out. However, these members are considered when calculating a maximum numbers of users allowed in the contact list.
Migration from Microsoft Office Communication Server	During migration from Microsoft Office Communication Server, the Enterprise Groups feature is not supported until users are fully migrated to the IM and Presence Service node.
LDAP synchronization	If you change the synchronization option in the **LDAP Directory Configuration window** while the synchronization is in progress, the existing synchronization remains unaffected. For example, if you change the synchronization option from **Users and Groups** to **Users Only** when the synchronization is in progress, the users and groups synchronization still continues.

Limitation	Description
Group search functionality over the Edge	Group search functionality over the Edge is offered in this release, but has not been fully tested. As a result, full support for group searches over the Edge cannot be guaranteed. Full support is expected to be offered in a future release.
Cisco Intercluster Sync Agent service periodic synchronization	If a group name or a group member name is updated in the external LDAP directory, it gets updated on the Cisco Jabber contact list only after the periodic Cisco Intercluster Sync Agent service synchronization. Typically, the Cisco Intercluster Sync Agent service synchronization occurs every 30 minutes.
Synchronization of users and user groups through different synchronization agreements in LDAP configuration	If users and user groups are synchronized into the Cisco Unified Communications Manager database as part of the same synchronization agreement, the user and group association gets updated as expected in Cisco Unified Communications Manager database after synchronization. However, if a user and user group are synchronized as part of different synchronization agreements, the user and the group may not get associated in the database after the first synchronization. The user and group association in the database depends on the sequence in which the synchronization agreements are processed. If the users are synchronized ahead of the groups, then the groups may not be available in the database for association. In such cases, you must ensure that the synchronization agreement with groups is scheduled ahead of the synchronization agreement with the users. Otherwise, after the groups synchronize into the database, the users will get associated with the groups after the next manual or periodic sync with the sync type set as Users and Groups. Users and corresponding group info will be mapped only when the agreement sync type is set as Users and Groups .
Tested OVA information for Enterprise Groups	**Tested Scenario** In a Intercluster deployment with two clusters Cluster A and Cluster B: Cluster A has 15K OVA and 15K users enabled for IM and Presence Service out of 160K users that are synced from Active Directory. The tested and supported average number of enterprise groups per user on 15K OVA cluster is 13 enterprise groups . Cluster B has 25K OVA and 25K users enabled for IM and Presence Service out of 160K users that are synced from Active Directory. The tested and supported average number of enterprise groups per user on 25K OVA is 8 enterprise groups. The tested and supported sum of user's personal contacts in roster and the contacts from enterprise groups that are in a user's roster is less than or equal to 200. **Note** In environments with more than 2 clusters these numbers are not supported.

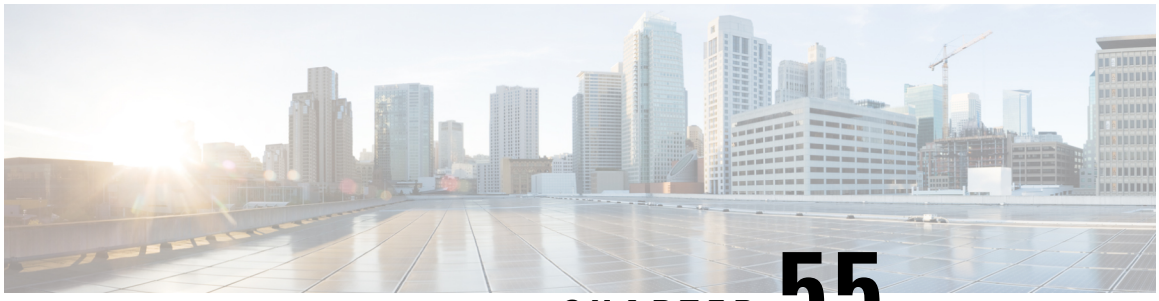
Limitation	Description
Export Contact List	When you export the user's contact list using **Bulk Administration** > **Contact List** > **Export Contact List**, the Contact List CSV file doesn't include the details of enterprise group they had in Jabber client.



PART XIII

Device Management

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- Headset Services, on page 693
- Native Phone Migration using IVR and Phone Services, on page 701
- Video Endpoints Management, on page 721



CHAPTER 55

Headset and Accessories Management

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- Feature Compatibility for Headset and Accessories Management, on page 675
- Workflow: Configure Headset Serviceability, on page 677
- Headset and Accessories Template Management, on page 681
- Firmware Management, on page 686
- Headset and Accessories Inventory Management, on page 687
- Headset and Accessories Troubleshooting and Diagnostics, on page 691

Headset and Accessories Management Overview

Headset and Accessories Management enhances your Cisco headset deployment, letting administrators manage headset serviceability from Cisco Unified Communications Manager. From Cisco Unified CM Administration, administrators can:

- Remotely configure headset settings such as wireless power range, audio bandwidth, and Bluetooth on/off.
- Define and control the headsets or accessories firmware.
- Get a detailed inventory of all the headsets and accessories in your deployment.
- Diagnose and troubleshoot headsets with Remote PRT, headset metrics in Call Management Records (CMR), and alarms.

Feature Compatibility for Headset and Accessories Management

Cisco Headset and Accessories Management is supported in Unified Communications Manager from the following releases:

- Release 12.5(1)SU4 for 12.x releases

Along with the Unified Communications Manager version, feature support is dependent on the firmware versions of Cisco Headsets and Accessories, Cisco IP Phone, and Cisco Jabber. The following table lists the available headset and accessories management features depending on the headset or accessories, phone, and Unified Communications Manager versions you use.

Note The Cisco Headset and Accessories Management feature is not supported in 12.0(x) or 12.5(1). For earlier versions, you may have a limited support for sending headset and accessories configuration templates for IP phones manually via the `defaultheadsetconfig.json` configuration file and TFTP. Refer to your headset Administration Guide for details.

Table 61: Headset Serviceability Features for Cisco IP Phones

Serviceability Feature	Unified CM 12.5(1) or earlier + Phone Firmware 12.1(1) or earlier	Unified CM 12.5(1)SU1 and above** + Phone Firmware 12.1(1) or earlier	Unified CM 12.5(1) or earlier + Phone Firmware 12.5(1)	Unified CM 12.5(1)SU1 and above** + Phone Firmware 12.5(1)	Unified CM 12.5(1) or earlier + Phone Firmware 12.5(1)SR3	Unified CM 12.5(1)SU1 and above** + Phone Firmware 12.5(1)SR3
Manual Remote Configuration	—	—	X	N/A	X	—
Headset Firmware Upgrade on Unified CM	—	—	—	—	—	X
Remote Headset and Accessories Configuration on Unified CM	—	—	—	—	—	X
Headset and Accessories inventory on Unified CM	—	—	—	—	—	X*
Configuration Reset on the phone UI	—	—	—	—	X	X
Headset Call Management Records (CMR)	—	—	—	—	—	X*

- * This feature is only available on headsets with Headset Firmware 1.5 or later.
- **This feature is not supported in the 12.0.x and 12.5(1) releases.
- N/A When you upgrade to Unified CM 12.5(1) or higher from an earlier version, most Cisco IP Phones will upgrade automatically to Phone Firmware 12.5(1)SR3 or higher versions.

Table 62: Headset Serviceability Features for Cisco Jabber

Serviceability Feature	Unified CM 12.5(1) or earlier + Jabber version 12.5(1) or earlier	Unified CM 12.5(1)SU1 and above** + Jabber version 12.5(1) or earlier	Unified CM 12.5(1) or earlier + Jabber version 12.6(1)	Unified CM 12.5(1) and above** + Jabber version 12.6(1)	Unified CM 12.5(1) or earlier + Jabber version 12.6(1)MR	Unified CM 12.5(1) and above** + Jabber version 12.6(1)MR
Headset Firmware Upgrade through Unified CM	—	—	—	—	—	X
Remote Headset and Accessories configuration through Unified CM	—	—	—	X	—	X
Headset and Accessories inventory on Unified CM	—	—	—	X*	—	X*
Local configuration reset	—	—	—	—	X	X
Local UI configuration	—	—	X	X	X	X
Local Headset and Accessories version display	—	—	—	—	X	X

- * This feature can only detect headsets with Headset Firmware 1.5 or later.
- **This feature is not supported in the 12.0.x and 12.5(1) releases.

Third-Party Headset and Accessories Support

If you are deploying third-party headsets or accessories, Unified Communications Manager supports headset and accessories inventory management with limited information for the third-party headsets or accessories right from the Cisco Unified CM Administration interface. Unified Communications Manager does not support headset or accessories configuration templates, firmware, diagnostics, and headset CMRs for third-party headsets.

Workflow: Configure Headset Serviceability

Use the following workflow to guide you through the setup of your Cisco Headset Serviceability feature.

After you complete this workflow, you can configure headset or accessories settings, maintain headset or accessories latest firmware loads, headset or accessories association to users, enable headset-based Extension Mobility, and maintain inventory.

Procedure

	Command or Action	Purpose
Step 1	Activate Cisco Headset Service, on page 678	Turn on **Cisco Headset Service** in Cisco Unified Serviceability.
Step 2	Prepare Your Headset COP Files, on page 679	Make sure you install and upgrade the latest headset/accessories firmware using a COP file.
Step 3	Configure User Profiles for Headset Users, on page 680	If you haven't yet configured User Profiles, use this procedure to set up profiles for your users. If all User Profiles are configured, you can skip this task.
Step 4	Apply User Profiles to End Users, on page 681	Assign User Profiles to your end users. If you've already assigned User Profiles, you can skip this task.
Step 5	Configure a Headset and Accessories Template, on page 685	Configure default settings and firmware for a Cisco headset and accessories template. Associate User Profiles to the template such that users whom use that User Profile are assigned to this headset and accessories template.
Step 6	View Headset and Accessories Inventory, on page 689	Check that you can see your deployed headset and accessories inventory through the Cisco Unified CM interface.

Activate Cisco Headset Service

Before you can begin administering Cisco Headsets and Accessories through the Cisco Unified CM Administration interface, turn on in Cisco Unified Communications Manager Serviceability.

Note Cisco Headset service should be activated on all the Unified Communications Manager nodes wherever Cisco CallManager service is already running. Ensure that you activate the Cisco Headset service on the Unified Communications Manager nodes where you want to administer headsets using the Cisco Unified CM Administration interface. The Cisco CallManager service will be automatically activated when you enable the Cisco Headset service. Deactivate the Cisco CallManager service if you do not need it.

Procedure

Step 1 From Cisco Unified CM Administration, navigate to **Cisco Unified Serviceability** and click **Go**.

Step 2 Select **Tools** > **Service Activation**.

Step 3 Check the **Cisco Headset Service** check box from the CM Services section and select **Save**.

What to do next

Prepare your Headset COP Files.

Prepare Your Headset COP Files

You can install and upgrade the latest headset firmware using a COP file. A headset COP file contains all the firmware versions of different headset models along with their configuration data.

Note Ensure that the Cisco Headset service is up and running before the COP file is installed.

Ensure that the headset COP file is installed on all nodes of Unified Communications Manager.

1. Install or upgrade the COP file to the Unified Communications Manager system before you can start using your Cisco headsets .

When you connect your headset to the endpoints, the headset template configuration changes are applied. If you make any updates to the headset template configurations on Unified Communications Manager, the endpoints apply these configuration updates on the connected headsets .

All configuration updates depend on the version of the headset template in the COP file. If the headset template version is higher in the latest COP file, the configuration file on Unified Communications Manager is updated. If the configuration file in the COP file is upgraded, the headset template version in Unified Communications Manager is updated irrespective of the version of the template and vice versa. The following list shows the various template version update scenarios after a COP file upgrade:

- If the Unified Communications Manager is currently installed with the headset template version 1-10 and you upgrade your Unified Communications Manager server that has headset template version 1-12, then the chosen headset template version is 1-12. Unified Communications Manager opts for the higher headset template version.
- If the Unified Communications Manager is currently installed with the headset template version 1-10 and you upgrade your Unified Communications Manager server that has headset template version 1-9, then the chosen headset template version is 1-10. Unified Communications Manager opts for the higher headset template version.
- If the Unified Communications Manager is currently installed with the headset template version 1-10 and you install a COP file that has headset template version 1-12, then the chosen headset template version is 1-12. Headset template installed with the COP files is the preferred option.
- If the Unified Communications Manager is currently installed with the headset template version 1-10 and you install a COP file that has headset template version 1-9, then the chosen headset template version is 1-9. Headset template installed with the COP files is the preferred option.
- If you had a COP file installed that has headset template version 1-12 and you upgrade your Unified Communications Manager server having headset template version 1-10, then the chosen headset template version is 1-12. Unified Communications Manager opts for the higher headset template version.

Configure User Profiles for Headset Users

If you haven't yet configured User Profiles for your users, use this procedure to set up profiles. Your headset templates will be assigned to users via their User Profile. If you've already configured User Profiles, you can skip this task.

Note Configure multiple User Profiles for different groups of users as per your deployment needs. By default, all User Profiles get assigned to the System default headset template. You can assign them to customized templates when you configure your headset template.

Procedure

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

Step 2 Click **Add New**.

Step 3 Enter a **Name** and **Description** for the user profile.

Step 4 Assign a **Universal Device Template** to apply to users' **Desk Phones**, **Mobile and Desktop Devices**, and **Remote Destination/Device Profiles**.

Step 5 Assign a **Universal Line Template** to apply to the phone lines for users in this user profile.

Step 6 If you want the users in this user profile to be able to use the self-provisioning feature to provision their own phones, do the following:

a) Check the **Allow End User to Provision their own phones** check box.

b) In the **Limit Provisioning once End User has this many phones** field, enter a maximum number of phones the user is allowed to provision. The maximum is 20.

Step 7 If you want Cisco Jabber users associated with this user profile to be able to use the Mobile and Remote Access feature, check the **Enable Mobile and Remote Access** check box.

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

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

Step 8 Assign the Jabber policies for this user profile. From the **Jabber Desktop Client Policy**, and **Jabber Mobile Client Policy** drop-down list, choose one of the following options:

- No Service—This policy disables access to all Cisco Jabber services.
- IM & Presence only—This policy enables only instant messaging and presence capabilities.
- IM & Presence, Voice and Video calls—This policy enables instant messaging, presence, voicemail, and conferencing capabilities for all users with audio or video devices. This is the default option.

Note
Jabber desktop client includes Cisco Jabber for Windows users and Cisco Jabber for Mac users. Jabber mobile client includes Cisco Jabber for iPad and iPhone users and Cisco Jabber for Android users.

Step 9 If you want the users in this user profile to set the maximum login time for Extension Mobility or Extension Mobility Cross Cluster through the Cisco Unified Communications Self Care Portal, check the **Allow End User to set their Extension Mobility maximum login time** check box.

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

Step 10 Click **Save**.

Apply User Profiles to End Users

Associate your users to the User Profiles that you've created. The User Profile must be associated with the end user, and the MAC of the device must be added under the controlled devices to apply the headset template configuration changes.



Note If you've already assigned all users to the appropriate User Profiles, you can skip this task.

Procedure

Step 1 To add a new end user to the Unified Communications Manager database manually, perform the following:

a) In Cisco Unified CM Administration, choose **User Management > End User**.
b) Click **Add New**.
c) Enter the **User ID** and **Last name**.
d) Choose the **User Rank** from the drop-down list.
e) Complete the fields in the **End User Configuration** window. For field descriptions, see the online help.
f) Click **Save**.

Step 2 To associate the end user with the device, perform the following:

a) In Cisco Unified CM Administration, choose **Device > Phone**.
b) Select the Cisco IP Phone or device.
c) Under Device Information, select **User** as the Owner and select the **Owner User ID**.
d) Click **Save** and **Apply Config** for the configuration changes to take effect.

Headset and Accessories Template Management

You can assign headset and accessories templates to user profiles in Cisco Unified Communications Manager to configure default headset settings for your users. The headset and accessories template provides the option to associate User Profiles. Unified Communications Manager supports the following types of headset and accessories templates:

Standard Default Headset Configuration Template

This is the system default template that contains the factory default settings for all headset and accessories model series. This template contains the headset or accessories settings supported by the latest headset or accessories firmware installed on your system for all your headset or accessories model series. You cannot edit the default settings though you can change the profile configuration setting.

Note The Standard Default Headset Configuration template is created only when the **Cisco Headset Service** is activated in the Cisco Unified Serviceability user interface.

By default, all User Profiles are associated to the standard headset template unless the administrator associates these user profiles to any of the custom defined headset templates. You can make copies of the standard default headset template to create custom template with customized values of the parameters including the headset or accessories firmware version.

System Generated Custom Headset Template

For some earlier releases that did not support the full Cisco Headset Serviceability feature, administrators could configure and deploy headset and accessories templates manually via the `defaultheadsetconfig.json` configuration file and TFTP. If you used this method on a previous release, and then upgrade to this release, the config file is converted to the **System Generated Custom Headset Template** and displays in the **Headset and Accessories Configuration Template** window. Following the upgrade, users and devices that used the config file are associated to this custom template.

Custom Headset Configuration Template

From Cisco Unified CM Administration, use the **Device** > **Headset and Accessories** > **Headset and Accessories Template** window to customize headset and accessories templates as per your deployment needs. You can assign different headset parameters to different models in the same template. You can also assign different firmware loads to different headset or accessories models. The custom headset or accessories settings can be assigned to specific sets of users by associating the User Profile(s) to the Custom Headset Template.

Table 63: Headset and Accessories Configuration Template Settings

Field	Description
Headset and Accessories Template Configuration	
Name	Enter a unique name to identify the headset and accessories template.
Description	Enter a description that identifies use of the template.
Model and Firmware Settings	
Choose Model Series	Choose any supported headset or accessories model that offers reliable, high-quality sound for your device.

Field	Description
Add	For a standard template, you can view the default pre-defined firmware versions and settings of the headset or accessories models. You cannot edit the default values. For customized templates, click **Add** to add a new headset or accessories model and corresponding settings. You cannot add another existing headset or accessories model in the same template. You can add different headset or accessories models in a customized template; however, you can only use one firmware per headset or accessories model. For more information on headset parameters, see the "Headset Configuration Parameters" table below. For Standard Default Headset Template Configuration, you can only edit settings by installing a headset COP file.
Firmware	Select the required firmware version. • Remain on current version—Choose this option if you want the headset or accessories to remain on the existing firmware version (that is, the headset or accessories firmware version is not upgraded to the latest firmware version on the system). • Latest—Choose this option if you want to upgrade the headset or accessories firmware version to the latest firmware version on the system.
Delete	For customized templates, click **Delete** to remove the headset or accessories model from the headset and accessories template.
Profile Configuration	
Available User Profiles	Lists the configured User Profiles that are available to use with this headset and accessories template. To associate a User Profile to this template, select the profile and click the down arrow to move the template to Assigned User Profiles. **Note** By default, all User Profiles get assigned to the Standard Default Headset Configuration Template. To associate a User Profile to a different template, create the new template and assign the User Profile to the new template.
Assigned User Profiles	Lists the User Profiles that will use this headset and accessories configuration template. For users assigned to this profile, the settings in this headset and accessories configuration template are applied to their Cisco headsets and accessories during registration. Click the arrows to add new User Profiles from the Available User Profiles list.

The following table describes the parameters in each headset and accessories template.



Note On-premises and multiplatform headset serviceability features are unavailable through an RJ-9 connection.

Table 64: Cisco Headset 500 Series Parameters

Parameter	Range	Default	Notes
Speaker Volume	0 – 15	7	Controls the level of sound in the headset. 0 is very low while 15 is loud. Configure this setting based on the ambient noise in the office environment.
Microphone Gain	Softer – Louder	Default	Gain controls how loud the user sounds to other people on the call. **Softer** means users sound quiet while **Louder** means users sound much louder. Configure this setting based on the ambient noise in the office environment.
Sidetone	Off – High	Low	Controls how much of a user's own voice they can hear through their headset.**Off** turns off the sidetone while **High** means that users receive much more feedback from their headset microphones.
Equalizer	Warmest – Brightest	Default	Controls the Equalizer settings. Warmer settings mean users hear more bass in their headsets, while a brighter setting means users hear more treble.
Audio Bandwidth	Wide Band, Narrow Band	Wide Band	Controls the Digital Enhanced cordless Telecommunications (DECT) codec in the Cisco Headset 560 Series. In a dense DECT environment, set the field to **Narrow Band** to limit the Cisco Headset 560 Series to the G.727 codec.
Bluetooth	On, Off	On	Controls the use of Bluetooth on the Cisco Headset 560 Series with Multibase. When this parameter is set to **Off**, the base deletes all devices paired with it.
Conference	On, Off	On	Controls the use of the conferencing feature on the Cisco Headset 560 Series. Conferencing allows up to three guest headsets to pair with the same base at once. See *Cisco Headset 500 Series User Guide* for more information on conferencing.

Parameter	Range	Default	Notes
Firmware Source	Allow from UCM or Cisco Cloud (firmware will upgrade only), Restrict to UCM only (firmware may upgrade or downgrade)	Allow from UCM or Cisco Cloud	Controls the headset's firmware upgrade source. By default, users can upgrade their headset through a devices and software connected to Unified CM or through a cloud-connected device or software. You can restrict your headsets to only accept firmware changes through a Unified CM source.
DECT Radio Range	Autorange, Medium Range, Short Range	Medium Range	Controls the maximum distance between the Cisco Headset 560 Series and its base. By default, the bases have a DECT range of over 330 feet (100 meters) in ideal conditions. If you configure the DECT radio range to **Medium Range** or **Short Range**, the headset base consumes less power but users can't move as far from the base while on a call. Configure DECT radio range to **Short Range** for high density headset deployment. For more detailed information on DECT deployment, refer to the white paper on Cisco Headset deployment, How to Deploy DECT at Work for the Cisco Headset 560 Series.
Headset dock behavior	On, Off	On	Controls how the Cisco Headset 560 Series behaves if you lift the headset off the base when you have an incoming call.

Configure a Headset and Accessories Template

Use this procedure to configure a headset and accessories template with customized settings that you can apply to Cisco Headsets and Accessories. You can create a customized template or use the system-defined Standard Default Headset Template.

Note The Standard Default Headset Configuration Template is a system-defined template. You can assign new User Profiles to the Standard Default Headset Template but you can't edit the template. By default, all user profiles are assigned to this template. To disassociate a user profile from this template, you must assign the profile to a new template.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device > Headset and Accessories > Headset and Accessories Template**.

Step 2 Do either of the following:

- To edit an existing template, select the template.
- To create a new template, select any existing template and click **Copy**. The existing settings are applied to your new template.

Step 3 Add a **Name** and **Description** for the template.

Step 4 Under **Model and Firmware Settings**, assign any customized headset or accessories settings that you want to apply to this template. To add a new setting, click the **Add** button and configure the settings.

Step 5 Use the up and down arrows to move the User Profiles that you want to assign to this template to the **Assigned Uers Profiles** list box. All users whom are assigned to those profiles will also be assigned to this headset and accessories template.

Step 6 Click **Save**.

Step 7 Use the **Set to Default** button to return to the default template settings.

Step 8 Click **Apply Config**.

For a Standard Default Headset Configuration Template, the **Apply Config** button takes effect for the following:

- Devices owned by users you added to the Assigned User Profile list
- Anonymous devices

For a Customized Headset Configuration Template, the **Apply Config** button takes effect only for devices owned by users you added to the **Assigned User Profiles** list.

Firmware Management

Most phones and devices connected to the Unified Communications Manager support the Cisco Headset 500 Series and Cisco Headset 700 Series. Install the latest phone firmware release and device package before connecting your headset or accessories to a phone. When the headset or accessories first connects, it downloads the required firmware and begins the upgrade process.

For a given headset or accessories model, the following two firmware options are supported:

- **Remain on current version**—Choose this option if you want the headset or accessories to remain on the existing firmware version (that is, the headset or accessories firmware version is not upgraded to the latest system firmware version).
- **Latest**—Choose this option to upgrade or downgrade the headset or accessories. The system installs and runs the chosen software, even if that firmware is an older release from what the headset or accessories currently has.

  For example, if you choose **1-5-1-10** as the latest, that firmware will be installed on the headset or accessories regardless of whether the headset or accessories currently has **1-5-1-9** or **1-5-1-11**.

Firmware Considerations

- Users assigned to the standard headset template will always receive the latest headset or accessories firmware and settings.

- Settings shown in the Headset Template Configuration (both Standard and Custom) are always set to the **Latest firmware** for all headset and accessories model series.

Headset and Accessories Inventory Management

Cisco IP Phones send headset and accessories inventory data to Unified Communications Manager whenever the headset and accessories are in a connected or disconnected state. Unified Communications Manager stores the inventory data so you can generate an Inventory Summary Report or Custom Inventory Report for all headsets and accessories deployed in this server.

Report information includes: headset or accessories serial and model number, docking station details, firmware, configuration templates used, vendor details, and headset or accessories connection status to devices.

Headset and Accessories Inventory

From Cisco Unified CM Administration, use the **Device** > **Headset and Accessories** > **Headset and Accessories Inventory** window to view a full list of all headsets and accessories that are deployed on your server. You can use this information to generate reports for all deployed headsets and accessories. If you click the Serial Number of the device, you can view details of individual headsets and accessories in a pop-up window.

Table 65: Headset and Accessories Inventory Settings

Field	Description
Serial Number	Serial Number of the headsets or accessories. This number is unique for every individual headset or accessories. **Note** For non-Cisco headsets or accessories, the Device Name is used as the Serial Number. Using the same non-Cisco headset or accessories with multiple phones creates duplicate headsets or accessories records. **Note** For information on how to locate the Serial Number for a specific headset or accessories, see the *Headset Administration Guide* for that headset or accessories model.
Model	Model number of the headset or accessories.
Vendor	Displays vendor details.
Type	Indicates the type of headset connection: Wired, DECT Wireless, or Unknown.
Firmware	Displays the most current firmware load of the headset or accessories.
User	Displays information of the end user using the phone or device.
Attached Phone Owner User ID	Displays information of the end user using the phone or device. The field is blank when there are no headsets or accessories associated.

Field	Description
Headset/Accessories Owner	Displays the end user information associated with the Serial Number of the headset or accessories.
Template	Display the name of the headset or accessories configuration template.
Status (since)	Displays the status of the headset or accessories activities. It can be: Connected or Disconnected.
Dock Model	Displays the type of docking model station.
Device Name	Name of the device to which the headset or accessories are connected to.
Device Model	Displays the Cisco IP Phone or Cisco Jabber model number. For example, CP-8865 is a Cisco IP Phone model. CSF is a device type for either Cisco Jabber for Mac or Cisco Jabber for Windows.
Software Version	Displays the latest version of the software used. It can be a phone firmware or a Jabber software version.
Headset/Accessories Age (days)	Displays the age of the headset or accessories. If the record is deleted, the headset or accessories age is reset.

Headset and Accessories Inventory Download

Important This section is applicable from Release 12.5(1)SU4 and Release 14 onwards.

In Cisco Unified Communications Manager Administration, under **Headset and Accessories > Headset and Accessories Inventory** menu path, select **Headset and Accessories Inventory Download** from the Related Links drop-down list to download detailed information of the headsets and accessories in the CSV File Format.

You can use this information to analyze data for use cases such as tracking the headsets and accessories usage, third-party headsets in your deployment, and Refresh headsets.

Headset and Accessories Inventory Management Task Flow

Procedure

	Command or Action	Purpose
Step 1	View Headset and Accessories Inventory, on page 689	Lists headsets and accessories deployed on the server.
Step 2	Associate Phone Owner as Headset or Accessories Owner, on page 689	Associates headsets or accessories to the users.

View Headset and Accessories Inventory

You can view a full list of all headsets and accessories deployed on your server. You can use this information to generate reports for all deployed headsets and accessories.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Headset and Accessories** > **Headset and Accessories Inventory**.

Step 2 Do either of the following:

- Select **Find** to see a full list of headsets deployed on your server.
- Enter a one or more search criteria into the search box and select **Find**.

Associate Phone Owner as Headset or Accessories Owner

Use this procedure to associate bulk headsets or accessories to the user.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Headset and Accessories** > **Headset and Accessories Inventory**.

Step 2 Click **Find** to view a full list of headsets or accessories deployed on the server.

Step 3 Click **Select All** or choose the required Serial Numbers and click **Associate Phone Owner as Headset/Accessories Owner** to associate headsets or accessories to the user.

Note
You can't associate headsets or accessories when they are already associated or doesn't have phone owners to associate with. The headset or accessories association is visible in the "Headset/Accessories Owner" column after the page reloads.

You can associate a maximum of only 15 headsets or accessories to a specific user. Once the maximum limit is reached for a specific user, the remaining headsets or accessories aren't associated, and an error is displayed.

Step 4 (Optional) Select the required Serial Numbers and click **Disassociate Headset /Accessories Owner** to dissociate the headset or accessories serial number from the selected user.

Note
You can't disassociate headsets or accessories if it's not associated with the headset or accessories owner.

Step 5 (Optional) To view end user configuration and headset or accessories association details, click the Username link in the **Attached Phone Owner User ID** or **Headset/Accessories Owner** column.

Note

The **End User Configuration** window displays the headset or accessories association and disassociation details.

Headset and Accessories Inventory Summary

From Cisco Unified CM Administration, you can use the **Device** > **Headset and Accessories** > **Headset and Accessories Inventory Summary** window to view an aggregate summary of your deployed headsets and accessories in the **Headset and Accessories Inventory Summary** window.

Headset and Accessories Inventory by Model

Field	Description
Headset/Accessories Model	The headset or accessories model number.
Quantity	Lists the number of headsets or accessories for each model type in your deployment. **Note** Click the link in the **Quantity** column to navigate to the detailed Headset and Accessories Inventory page, filtered by model type.

Headset and Accessories Inventory by Status

Click the hyperlinks in the **Headset/Accessories Model**, **Active**, **Inactive**, or **Unassigned** columns to navigate to the detailed Headset and Accessories Inventory page for each status.

Field	Description
Headset/Accessories Model	The headset or accessories model number.
Active	The headset or accessories has connected within the last 30 days.
Inactive	The headset or accessories hasn't connected in the last 30 days.
Unassigned	The user ID doesn't exist in the system or the inventory record doesn't have a user ID mapping.

Get an Aggregate Summary of Your Deployed Headsets and Accessories

You can view an aggregate summary of your deployed headsets and accessories in the **Headset and Accessories Inventory Summary** window.

Procedure

In Cisco Unified CM Administration, select **Device** > **Headset and Accessories** > **Headset and Accessories Inventory Summary**.

You can view a breakdown of headset and accessories inventory by model or by headset and accessories status.

Headset and Accessories Troubleshooting and Diagnostics

You can configure Unified Communications Manager or Cisco Unified Real-Time Monitoring Tool (RTMT) to collect Problem Report Tool (PRT) logs for headsets or accessories connected to Cisco IP Phones. The PRT includes data on call quality, codecs used, audio settings, wireless settings, and alert logs.

Unified Communications Manager stores the call diagnostics details for Headsets and Accessories. Cisco IP Phones send headset or accessories diagnostics data in Headset-Stat header either in a BYE message or a 200 OK response to BYE message to update the CMRs in Unified Communications Manager.

Cisco IP Phones share the headset and accessories diagnostics data with Unified Communications Manager and this information is stored in the following fields in the CMR record:

- SN—Serial number of the headset or accessories.
- Metrics—Headset and accessories metrics such as RSSI frame errors, connection drop reason, beacon moves, audio settings, and DECT bandwidth.

For detailed information on how to export and view CMR records, see the *Call Reporting and Billing Administration Guide for Cisco Unified Communications Manager*.

Note Headset CMR records apply to Cisco Headset 500 Series, but not to 700 Series.

Generate PRT for Endpoints on Unified CM

Use this procedure to trigger the Problem Reporting Tool (PRT) on the endpoints.

Procedure

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

Step 2 Click **Find** and select one or more phones that the headset connects to.

Step 3 Click **Generate PRT for Selected** to collect PRT logs for the headsets used by the selected phones.

Step 4 Click **Save**.

Cisco Unified Communications Manager sends SIP Notify messages to remotely trigger the log collection on the phone and upload it to the log server configured in the "Customer support upload URL" parameter.

Generate PRT for Endpoints on RTMT

Devices or endpoints generate alarms for each critical event for diagnostics and troubleshooting. These alarms are generated using the Problem Reporting Tool (PRT) available in the Trace Collection menu or the Device Monitoring menu of the Cisco Unified Real-Time Monitoring Tool (RTMT) user interface.

Procedure

Step 1 Open the Trace and Log Central options.

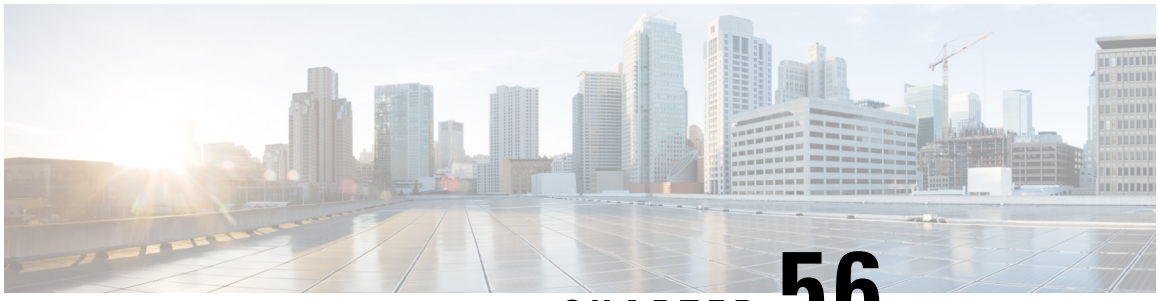
Step 2 In the Trace & Log Central tree hierarchy, choose **Generate PRT**.
The Generate PRT wizard appears.

Step 3 Enter the Device name as configured in the Find and List Phones page in the Cisco Unified CM Administration user interface.

Step 4 Click **Generate PRT**.

The generated report is uploaded at the **Customer support upload URL**. The download option is available only if the **Customer support upload URL** parameter is configured at the Enterprise, Profile, or Device level in the Cisco Unified CM Administration user interface.

Note
Check the **Customer support upload URL** parameter in the Enterprise, Profile, or Device level configuration page settings. Else, PRT generation fails.



CHAPTER 56

Headset Services

- Headset Services Overview, on page 693
- Headset Services Prerequisites, on page 694
- Headset Services Administrator Configuration Task Flow, on page 694
- Headset Services End User Association Task Flow, on page 698

Headset Services Overview

Headset Services allow you to connect the Cisco Headset into its supported Devices to provide simple and integrated user experiences such as Headset-based Extension Mobility and many more in the future.

Headset-based Extension Mobility is the first feature introduced under the Headset Services. When you connect your Cisco headset to Extension Mobility enabled devices, it provides a seamless login experience for extension mobility log in and log out.

Headset Services allow the administrator and end user to associate headset(s) from any devices such as a self-owned device, shared space, and common area device. This association helps in authentications and creating a customized experience for its users. This feature supports both wired and wireless headsets.

Headset association assigns the user's identity to the headset. You can log in to the services that need user identity.

The Unified Communications Manager interface allows the administrator to:

- Associate and disassociate headset to end users along with serial numbers.
- Enable Headset-based Extension Mobility.
- Import and export bulk users to headsets association.

Note Headset-based Extension Mobility login is not supported for Extension Mobility Cross Cluster (EMCC).

Headset-based Extension Mobility login works for devices supporting Mobile and Remote Access (MRA). The compatible Phone Firmware version is 14.1(1).

Headset-based Extension Mobility login does not work if the same user ID is controlling both the Headset and the Phone.

Headset Services Prerequisites

- Make sure that end users are already created in the Unified Communications Manager.
- For Extension Mobility login using the headset, ensure that Extension Mobility is enabled in the user device. And also, **Allow headset for Extension Mobility sign in and sign out** option is enabled so that the user can perform Extension Mobility log in or log out using Cisco headset.

Note Headset-based Extension Mobility feature supports only the latest firmware of 88XX and 78XX series Cisco IP Phones.

Headset Services Administrator Configuration Task Flow

The administrator can use the following task to associate headset to users and enable Headset-based Extension Mobility.

Procedure

	Command or Action	Purpose
Step 1	Headset Association to a User, on page 694	Specifies how to associate and dissociate the serial number with a user.
Step 2	Manage End User Headset Association, on page 695	**Optional**: Allows the end users to create a headset association for the devices.
Step 3	Enable Headset-based Extension Mobility, on page 696	Enables Extension Mobility for headsets from the Unified Communications Manager.
Step 4	Enable Pinless Extension Mobility Login, on page 696	Enables pinless Extension Mobility login.
Step 5	Configure Extension Mobility Headset Logout Timer, on page 697	Configures automatic logout timeout settings for headsets.

Headset Association to a User

Use this procedure to associate the headset with a user.

Procedure

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

Step 2 Click **Find** and choose an existing user to whom you want to associate the headset.

Step 3 In the **Associated Headsets** section, enter the serial number of the headset that you want to assign.

Step 4 Click **Save**.

Step 5 Click **(+)** if you want to associate more headsets to the selected user.

Note
You can associate a maximum of only 15 headsets to a specific user. The headset serial number is unique for every individual headset. The same headset can't be associated with two users. To move the headset association to a different user, you must first disassociate the headset from the first user.

For information on how to locate the Serial Number for a specific headset, refer to the *Headset Administration Guide* for that headset model.

Step 6 (Optional) Click **(-)** to dissociate the headset serial number for the selected user.

Step 7 Click **View Details** link to view the inventory details of a headset. For more information, see the Headset Inventory Settings section in the 'Headset and Accessories Management' chapter to view the headset details.

Manage End User Headset Association

Optional: Use this procedure to configure settings in the Unified Communication Manager to enable end users to associate headset using the **Headset Association** menu option on the device screen.

Procedure

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

Step 2 In the **Enterprise Parameters Configuration** section, choose one of the following to associate the end user headset to the device:

- Choose **Prompt user to initiate Headset Association from all devices** to display the **Headset Association** screen when the headset is connected to the device for the first time. By default, this parameter value is selected.
- Choose **Prompt user to initiate Headset association only from Extension Mobility-enabled devices** for the **Headset Association** screen to only appear in the Extension Mobility enabled devices.
- Choose **Do not prompt user to initiate Headset association from all devices** to disable **Headset Association** screen on all devices. This setting doesn't prevent the user from initiating headset association manually from the device menu.

Note
Any changes in the settings is not applicable to the headsets that are already associated with the end user.

Step 3 Click **Save** and **Apply Config** for the configuration changes to take effect.

Tip
Click the parameter name or the question mark **(?)** icon in the **Enterprise Parameter Configuration** window for detailed descriptions.

Enable Headset-based Extension Mobility

Use this procedure to allow users to log in to Extension mobility from associated headsets.

Before you begin

Ensure that you configure Cisco IP Phones and device profiles to the extension mobility service that headset users can log in, use, and log out of Extension Mobility using a headset. For more information, see Subscribe to Extension Mobility, on page 408.

Procedure

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

Step 2 From the **Server** field, choose the node running the Cisco Extension Mobility service.

Step 3 From the **Service** field, choose **Cisco Extension Mobility**.

Step 4 In the **Headset-based Extension Mobility** field, choose one of the following to use an associated headset for extension mobility login:

- Choose **Allow headset for Extension Mobility sign in and sign out** to allow the headset user to sign in with extension mobility. By default, this parameter value is selected.
- Choose **Do not Allow headset for Extension Mobility sign in and sign out** to restrict the headset user to sign in with Extension Mobility. If you choose this option, end users don't see the Extension Mobility login or logout screen when they connect their headsets.

Step 5 Click **Save**.

Enable Pinless Extension Mobility Login

Use this procedure for pinless Extension Mobility login using a headset associated with a user.

Note This feature is supported from release 12.5(1)SU3 onwards.

Before you begin

Specify the maximum duration the system waits for user input before automatically signing into the Extension Mobility profile in the **Service Parameter Configuration** > **Auto login timer after headset connect (seconds)** field.

Note The specified maximum duration takes effect only when **PIN entry for headset-based sign in** field is set to **Not Required**.

Procedure

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

Step 2 From the **Server** field, choose the node running the Cisco Extension Mobility service.

Step 3 From the **Service** field, choose **Cisco Extension Mobility**.

Step 4 In the **PIN entry for headset-based sign in** field, choose one of the following to enable or disable pinless Extension Mobility login:

- Choose **Required** to prompt the user to enter a PIN for Extension Mobility login. By default, this parameter value is selected.
- Choose **Not Required** to automatically sign in the user within a minute to Extension Mobility. The user isn't prompted to enter PIN details on the Phone UI.

Important
When the user automatically signs out within the set time or manually logged out using wired or wireless headset, we recommend the user to click **Cancel** to avoid automatic sign-in within the specified duration.

Step 5 Click **Save**.

Configure Extension Mobility Headset Logout Timer

Use this procedure to configure the auto-logout timeout settings.

Note If the Headset-based Extension Mobility service parameter in **Service Parameter Configuration** window is set to **Do not Allow headset for Extension Mobility sign in and sign out**, then configuring auto-logout timer value has no effect.

Procedure

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

Step 2 From the **Server** field, choose the node running the Cisco Extension Mobility service.

Step 3 From the **Service** field, choose **Cisco Extension Mobility**.

Step 4 In the **Auto logout timer after headset disconnect** field, enter the maximum duration value that the system can wait for user input when the headset is disconnected from the device before automatically logging out the user.

Note
By default, the value is set for 5 minutes. You can set the maximum value to 15 minutes.

Step 5 Click **Save**.

Headset Services End User Association Task Flow

The end user can use the following task to associate the headset and use the associated identity to login using Extension Mobility.

Procedure

	Command or Action	Purpose
Step 1	Associate a User Headset, on page 698	Creates a headset association to the end user.
Step 2	Skip Headset Association, on page 699	Allows skipping the headset association for the particular end user.
Step 3	Extension Mobility Login Using Headset, on page 699	Enables customized experience to use an associated headset for Extension Mobility login.
Step 4	Logout User from Extension Mobility Using Headset, on page 700	Helps to logout the headset from Extension Mobility within the default set time.

Associate a User Headset

Use this procedure to assign the headset to a user.

Procedure

Step 1 Connect the headset to the Cisco IP Phone.

The *Associate headset to user* popup screen appears on the IP Phone screen.

The Username is automatically populated if the device is in a shared space or common area or user is associated with the device. If the device is anonymous, the **User ID** field is blank and any end user can associate the headset providing the user credentials.

Step 2 Enter or modify **User ID** and **PIN**. Contact the administrator if you don't know the credentials.

Step 3 Click **Submit**.

Successful associating of headset message is displayed along with the Username.

If you enter invalid credentials (**User ID** or **PIN**) more than three times, the Cisco IP Phone displays an error message.

If the headset association fails, you can disconnect and connect in the headset to provide valid credentials or contact the administrator.

Step 4 (Optional) To associate headset manually through Cisco IP Phone, choose **Settings** > **Accessories** > **Cisco Headset Setup** > **Associate User**.

Note

The **Associate User** option is greyed out if the headset is disconnected. To enable, connect in the headset to the device.

Skip Headset Association

Use this procedure to skip associating the headset to a user.

Procedure

Step 1 Connect the headset to the Cisco IP Phone.

Step 2 Click **Exit** before associating the headset to a user.

Step 3 Click **Yes** if you don't want to associate the headset.
The headset association screen isn't prompted on any further connections to the device. If the same headset is connected to another device, the **Associate Headset to User** popup screen appears on the Cisco IP Phone screen and navigates you through the association process.

Extension Mobility Login Using Headset

Use this procedure to log in with Extension Mobility using a headset that has a user associated.

Procedure

Step 1 Connect the headset to the Cisco IP Phone.

Step 2 If the headset isn't associated, perform the following:

a) Enter the **User ID** and **PIN** to associate the headset with the user.
b) Click **Submit**.

The login screen displays a success message with the associated User ID and allows the user to sign in with Extension Mobility.

c) Click **Sign In** to complete the Extension Mobility login.

Step 3 If the headset is already associated with the user, perform the following:

a) Enter the **PIN** to login with Extension Mobility.
b) Select the required user profile.
c) Click **Submit**.

Step 4 If a user is already logged in to Extension Mobility on the device and another user plugs in a previously associated headset, the logout screen appears and allows the user to sign out previously logged in user.

Step 5 Click **Yes** to log out of the earlier profile.

Step 6 Enter the **PIN** to login with Extension Mobility.

Step 7 Click **Submit**.

Note
The phone resets each time the device profile changes, and the user profile is changed to the original profile.

Logout User from Extension Mobility Using Headset

Use this procedure to sign out the headset from the Extension Mobility enabled device.

Procedure

Step 1 Disconnect the headset from the Cisco IP Phone.

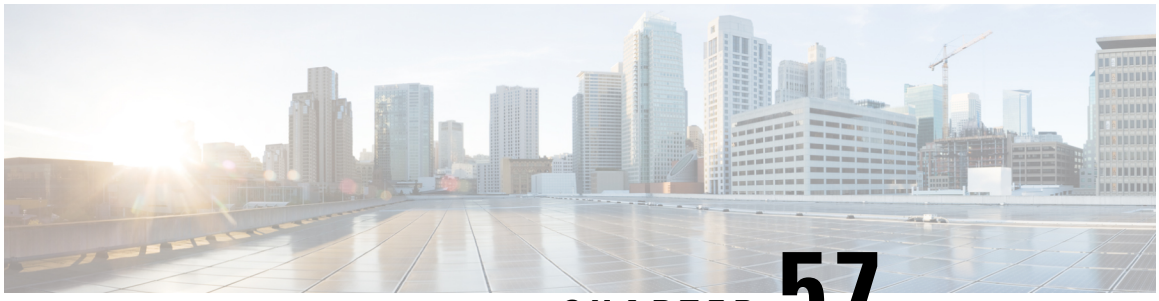
Step 2 Click **Sign Out**.

Note
The phone resets and the device profile is changed to the original device profile.

If you disconnect the headset during an ongoing call (one to one call or conference call), the call is not terminated, and the Extension Mobility sign out occurs only when the call ends.

You are automatically signed out within the set time if you haven't manually logged out or go out of range for a wireless headset. By default, the set time is 5 minutes. For more information, see the Configure Extension Mobility Headset Logout Timer, on page 697 section.

Step 3 Click **Cancel** if you want to retain the current Extension Mobility session. Reconnect within the default set time to retain the user profile and avoid a reset.



CHAPTER 57

Native Phone Migration using IVR and Phone Services

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- Phone Migration Prerequisites, on page 704
- Phone Migration Task Flow Using Self-Provisioning IVR, on page 705
- Phone Migration Task Flow Using Phone Migration Service, on page 710
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Native Phone Migration using IVR and Phone Services Overview

The Phone Migration feature is an easy and intuitive Cisco IP Phone migration solution native to Unified Communications Manager. It minimizes the cost and complexity of replacing deprecated or faulty phones. Using the solution, an end user or an administrator can easily migrate all the settings from an old phone to a new phone with a simple user interface. Solution supports following methods for migrating the phones:

- **Using Self-provisioning IVR Service**
- **Using Phone Migration Service**
- **Using Cisco Unified CM Administration Interface**

Following table provides a quick comparison of the various phone migration options:

Table 66: Different Phone Migration Options and Considerations

	Using Self-provisioning IVR Service	Using Phone Migration Service	Using Unified CM Administration Interface
End user or administrator driven phone migration	End user (Self-service)	End user (Self-service)	Administrator
Auto-registration required	Yes	No	No

	Using Self-provisioning IVR Service	Using Phone Migration Service	Using Unified CM Administration Interface
Migration steps	• Auto register a new phone • Dial self-provisioning IVR number • Follow the voice prompts	• Plug-in new phone to the network • Key in primary extension and PIN (optional)	• Sign in to Cisco Unified CM Administration interface • Choose "Migrate Phone" option in the Phone Configuration page of the old phone • Enter phone type (model & protocol) and MAC address of the new phone
Administrator involvement	Medium	Low	High

Note While phone migration using Self-provisioning IVR Service and Phone Migration Service facilitates the migration of the phone by an end user as a self-service; administrators, can use these methods to migrate the phone on-behalf of an end-user or common phones (for example, a lobby phone).

Note During phone migration, if the end user does not remember the PIN, administrators should advise the end user to log in to the Self Care Portal to change the PIN, if required.

Enterprise Parameters for Phone Migration

Migration depends on the following two parameters in the Enterprise Parameters Configuration page:

- **When Provisioning a Replacement Phone for an End User**—You can choose either **Retain Existing Phone(s)** (default option) or **Delete the Existing Phone for that End User**. If the **Retain Existing Phone(s)** option is selected, then during migration the old phone is marked as migrated and can be filtered in the Find and List Phones page to generate migration report.

Note If the administrator decides to choose the **Retain Existing Phone(s)** option for phone migration, a total of two licenses will be consumed, that is, one license each for the existing phone and the new phone picked up for migration.

Note During phone migration, if the administrator decides to choose the **Retain Existing Phone(s)** option for phone migration, the intercom DN is not migrated to the new device. If the **Delete the Existing Phone for that End User** option is selected, the intercom DN information is also migrated.

- **Security Profile for Migrated Phone**—This option determines the type of security profile that is applied for a migrated phone (old phone was in secure mode) during phone migration. Choosing the **Non-secure** profile brings up your device in the non-secure mode.
- **Phone Migration User Identification Prompt**—This parameter determines whether a user is able to proceed with phone migration using the Self-Service User ID or using the Primary Extension. If the **Use Enduser Self-Service User ID** option is selected, the end user is prompted to enter the unique Self-Service User ID before proceeding with phone migration. If the **Use Enduser Primary Extension** option is selected, you can migrate your phone after entering the primary extension number. In this mode, if the same DN exists in different route partitions, phone migration is not possible using IVR or Phone Migration service.

  The default value is **Use Enduser Primary Extension**.

To facilitate secure phone migration, the following Standard Universal Phone Security Profiles templates have been added. Based on the old phone security profile, the new phone chooses one of these new profiles.

- Universal Device Template - Security Profile - Non-Secure
- Universal Device Template - Security Profile – Authenticated
- Universal Device Template - Security Profile – Encrypted
- Universal Device Template - Security Profile - Encrypted - Device_TFTP
- Universal Device Template - Security Profile - Encrypted EC preferred

The following table lists the new devices security profile mappings after phone migration:

Table 67: Phone Security Profile Migration Data

Old Security Profile Settings				**New Mapped Profile Name**
Device Security Mode	**TFTP Encrypted Configuration**	**Transport Type**	**Key Order**	
Non-Secure	No	TCP	RSA	Universal Device Template - Security Profile - Non-Secure
Authenticated	No	TLS	RSA	Universal Device Template - Security Profile – Authenticated

Old Security Profile Settings				New Mapped Profile Name
Encrypted	No	TLS	RSA	Universal Device Template - Security Profile – Encrypted
Encrypted	Yes	TLS	RSA	Universal Device Template - Security Profile - Encrypted - Device_TFTP
Encrypted	Yes	TLS	EC preferred, RSA backup	Universal Device Template - Security Profile - Encrypted - Device_TFTP
Encrypted	No	TLS	EC preferred, RSA backup	Universal Device Template - Security Profile - Encrypted EC preferred

Phone Migration Prerequisites

Using Self-provisioning IVR

Before your end users can use self-provisioning for migration, the following should be configured:

- Enable Auto-registration.
- End users must have a primary extension. Ensure that the primary DN is always Line 1 on the phone or device.
- End users must be associated to a user profile or feature group template that includes a universal line template, universal device template and which has Self-Provisioning enabled.
- Ensure that the right “CTI Route Point” and “Application User” configurations are selected.
- Enable Self-Provisioning IVR service.

Using Phone Migration Service

Before your end users can use Phone Migration Service for migration, the following should be configured:

- Disable Auto-registration.
- End users must have a primary extension. Ensure that the primary DN is always Line 1 on the phone or device.
- Supported Phone Models: 88XX, 78XX, 8832, and 7832.
- Minimum supported phone version is Release 12.8.1 and above.

Phone Migration Task Flow Using Self-Provisioning IVR

Use the following task flow to guide you through the phone migration procedures.

After you complete this workflow, you can configure Self-Provisioning IVR service, migrate old or faulty Cisco IP Phones, and track the migrated phones list.

Procedure

	Command or Action	Purpose
Step 1	Activate Services for Self-Provisioning, on page 705	Activate the **Self-Provisioning IVR** and **CTI Manager** services in Cisco Unified Serviceability.
Step 2	Enable Autoregistration for Self-Provisioning, on page 706	Enable autoregistration parameter for self-provisioning.
Step 3	Configure CTI Route Point, on page 706	Configure a CTI route point to handle the self-provisioning IVR service.
Step 4	Assign a Directory Number to the CTI Route Point, on page 707	Configure the extension that users dial in order to access the self-provisioning IVR and associate that extension to the CTI route point.
Step 5	Configure Application User for Self-Provisioning, on page 707	Configure an application user for the self-provisioning IVR. Associate the CTI route point to the application user.
Step 6	Configure the System for Self-Provisioning, on page 708	Configure Self-Provisioning system settings.
Step 7	Enable Self-Provisioning in a User Profile, on page 709	Enables the users to Self-Provision phones in the user profile to which they are assigned.
Step 8	Migrate phones using any of these procedures: • Migrate Phones Using Self-Provisioning IVR (Administrator), on page 709 • Migrate Phones Using Self-Provisioning IVR (Phone Users), on page 710	Choose the migration procedure that applies for you. The Self-Provisioning IVR can be used by either administrators or phone users to migrate phones.
Step 9	View Phone Migration Report, on page 715	Following the migration, view a report that shows Cisco IP Phones that are migrated.

Activate Services for Self-Provisioning

Use this procedure to activate the services that support the Self-Provisioning feature. Ensure that both the Self-Provisioning IVR and Cisco CTI Manager services are running.

Procedure

Step 1 From Cisco Unified Serviceability, choose **Tools** > **Service Activation**.
Step 2 From the **Server** drop-down list, select the publisher node and click **Go**.
Step 3 Under **CM Services**, check **Cisco CTI Manager**.
Step 4 Under **CTI Services**, check **Self Provisioning IVR**.
Step 5 Click **Save**.

Enable Autoregistration for Self-Provisioning

Use this procedure for self-provisioning, you must configure the auto-registration parameters on the publisher.

Procedure

Step 1 In Cisco Unified CM Administration, choose **System** > **Cisco Unified CM**.
Step 2 Click on the publisher node.
Step 3 Select the **Universal Device Template** that you want to be applied to provisioned phones.
Step 4 Select the **Universal Line Template** that you want to be applied to the phone lines for provisioned phones.
Step 5 Use the **Starting Directory Number** and **Ending Directory Number** fields to enter a range of directory numbers to apply to provisioned phones.
Step 6 Uncheck the **Auto-registration Disabled on the Cisco Unified Communications Manager** check box.
Step 7 Confirm the ports that will be used for SIP registrations. In most cases, there is no need to change the ports from their default settings.
Step 8 Click **Save**.

Configure CTI Route Point

Us this procedure to configure a CTI Route Point for the Self-Provisioning IVR.

Procedure

Step 1 From Cisco Unified CM Administration, choose, **Device** > **CTI Route Points**.
Step 2 Complete either of the following steps:
a) Click **Find** and select an existing CTI route point.
b) Click **Add New** to create a new CTI route point.
Step 3 In the **Device Name** field, enter a unique name to identify the route point.
Step 4 From the **Device Pool** drop-down list, select the device pool that specifies the properties for this device.

Step 5 From the **Location** drop-down list, select the appropriate location for this CTI route point.

Step 6 From the **Use Trusted Relay Point** drop-down list, enable or disable whether Unified Communications Manager inserts a trusted relay point (TRP) device with this media endpoint. The default setting is to use the Common Device Configuration setting that is associated to this device.

Step 7 Complete the remaining fields in the **CTI Route Point Configuration** window. For more information on the fields and their settings, see the online help.

Step 8 Click **Save**.

Assign a Directory Number to the CTI Route Point

Use this procedure to set up the extension that users will dial in to access the self-provisioning IVR. You must associate this extension to the CTI route point that you want to use for self-provisioning.

Procedure

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

Step 2 Click **Find** and select the CTI route point that you set up for self-provisioning.

Step 3 Under **Association** click **Line [1] - Add a new DN**.
The **Directory Number Configuration** window displays.

Step 4 In the **Directory Number** field, enter the extension that you want users to dial to access the Self-Provisioning IVR service.

Step 5 Click **Save**.

Step 6 Complete the remaining fields in the **Directory Number Configuration** window. For more information with the fields and their settings, see the online help.

Step 7 Click **Save**.

Configure Application User for Self-Provisioning

You must set up an application user for the self-provisioning IVR and associate the CTI route point that you created to the application user.

Procedure

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

Step 2 Perform either of the following steps:

a) To select an existing application user, click **Find** and select the application user.

b) To create a new application user, click **Add New**.

Step 3 In the **User ID** text box, enter a unique ID for the application user.

Step 4 Select a **BLF Presence Group** for the application user.

Step 5 Associate the CTI route point that you created to the application user by performing the following steps:

a) If the CTI route point that you created does not appear in the **Available Devices** list box, click **Find More Route Points**.
The CTI route point that you created displays as an available device.

b) In the **Available Devices** list, select the CTI route point that you created for self-provisioning and click the down arrow.
The CTI route point displays in the **Controlled Devices** list.

Step 6 Complete the remaining fields in the **Application User Configuration** window. For help with the fields and their settings, see the online help.

Step 7 Click **Save**.

Configure the System for Self-Provisioning

Use this procedure to configure your system for self-provisioning. Self-provisioning provides users in your network with the ability to add their own desk phone through an IVR system, without contacting an administrator.

Note In order to use the self-provisioning feature, your end users must also have the feature enabled in their user profiles.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **Self-Provisioning**.

Step 2 Configure whether you want the self-provisioning IVR to authenticate end users by clicking one of the following radio buttons:

- **Require Authentication**—In order to use the self-provisioning IVR, end users must enter their password, PIN, or a system authentication code.
- **No Authentication Required**—End users can access the self-provisioning IVR without authenticating.

Step 3 If the self-provisioning IVR is configured to require authentication, click one of the following radio buttons to configure the method whereby the IVR authenticates end users:

- **Allow authentication for end users only**—End users must enter their password or PIN.
- **Allow authentication for users (via Password/PIN) and Administrators (via Authentication Code)**—End Users must enter an authentication code. If you choose this option, configure the authentication code by entering an integer between 0 and 20 digits in the **Authentication Code** text box.

Step 4 In the **IVR Settings** list boxes, use the arrows to select the Language that you prefer to use for IVR prompts. The list of available languages depends on the language packs that you have installed on your system. Refer to the Downloads section of cisco.com if you want to download additional language packs.

Step 5 From the **CTI Route Points** drop-down list, choose the CTI route point that you have configured for your self-provisioning IVR.

Step 6 From the **Application User** drop-down list, choose the application user that you have configured for self-provisioning.

Step 7 Click **Save**.

Enable Self-Provisioning in a User Profile

In order for users to be able to Self-Provision phones, the feature must be enabled in the user profile to which they are assigned.



Note If you don't know which user profile your users are using, you can open a user's settings in the End User Configuration window and view the **User Profile** field to get the correct profile.

Procedure

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

Step 2 Click **Find and select the user profile** to which the user is assigned.

Step 3 Assign **Universal Line Templates** and **Universal Device Templates** to the user profile.

Step 4 Configure user settings for Self-Provisioning:

- Check the **Allow End User to Provision their own phones** check box.
- Enter a limit for the number of phones a user can provision. The default is 10.
- If you want users to be able to use self-provisioning to reassign a previously assigned phone, check the **Allow Provisioning of a phone that is already assigned to a different End User** setting in the user profile page associated with the end user of old device. Users can reassign a previously assigned phone only if this check box is enabled in the User Profile that is associated to the old device.

Step 5 Click **Save**.

Phone Migration Tasks

After self-provisioning Authentication is setup, use any of the following procedures to migrate phones.

Migrate Phones Using Self-Provisioning IVR (Administrator)

Administrators can use this procedure to migrate Cisco IP Phones on behalf of an end user, or to migrate common phones (for example, a lobby phone).

Before you begin

Make sure that the old phone is in the "Unregistered" state before you proceed with migration. You can plug the new phone into the network, wait until the phone registers and then perform the migration tasks. Once the migration is successful, the device will re-register with the users phone configuration data.

Procedure

Step 1 Dial the extension that is assigned to the Self-Provisioning IVR from a new phone.

Step 2 Press **2** to replace an existing phone.

Step 3 Enter the primary extension number of an end user phone or common phone followed by the pound key (#).

Step 4 Enter the **Authentication Code** followed by the pound key (#).

Migration starts after a successful authentication. Following the migration, the phone restarts with configuration settings migrated from end user's old phone.

Migrate Phones Using Self-Provisioning IVR (Phone Users)

Phone users can use this procedure to migrate to a new Cisco IP Phone.

Before you begin

Make sure that the old phone is in the "Unregistered" state before you proceed with migration. You can plug the new phone into the network, wait until the phone registers and then perform the migration tasks. Once the migration is successful, the device will re-register with the users phone configuration data.

Procedure

Step 1 Dial the extension that is assigned to the Self-Provisioning IVR from a new Cisco IP Phone.

Step 2 Press **2** to replace an existing phone.

Step 3 Enter the primary extension number of your phone followed by the pound key (#).

Step 4 Enter your PIN followed by the pound key (#).

Migration starts after a successful authentication. After successful migration, the phone restarts with the configuration settings migrated from your old phone.

Note
If the phone is assigned to another user, a phone user can re-provision the phone provided an administrator has enabled the **Allow Provisioning of a phone already assigned to a different End User** option in the user's User Profile window. Talk to your administrator about this option.

Phone Migration Task Flow Using Phone Migration Service

Use the following task flow to guide you through the phone migration procedure using Phone Migration Service.

After you complete this workflow, you can migrate old or faulty Cisco IP Phones, and track the migrated phones list.

Procedure

	Command or Action	Purpose
Step 1	Disable Autoregistration, on page 711	Disable the autoregistration parameter before phone migration.
Step 2	Set Up Default Phone Load, on page 711	Configure the default phone load before the phone migration service.
Step 3	Configure Self-Provisioning Authentication, on page 712	Configure the required Self-Provisioning authentication settings.
Step 4	Migrate phones using any of these procedures: • Migrate Phones Using Phone Migration Service (Administrator), on page 712 • Migrate Phones Using Phone Migration Service (Phone Users), on page 713	Choose the migration procedure that applies for you. The Phone Migration Service can be used by either administrators or phone users to migrate phones.
Step 5	View Phone Migration Report, on page 715	Following the migration, view a report that shows Cisco IP Phones that are migrated.

Disable Autoregistration

To use the Phone Migration Service, autoregistration must be disabled.

Procedure

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

Step 2 Click on the publisher node.

Step 3 Check the **Auto-registration Disabled on the Cisco Unified Communications Manager** check box.

Step 4 Confirm the ports that will be used for SIP registrations. In most cases, there is no need to change the ports from their default settings.

Step 5 Click **Save**.

Set Up Default Phone Load

Use this procedure to set up the default phone load before the phone migration service.

Procedure

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

Step 2 Select the required phone model for migration.

Step 3 Enter the phone load and click the **Swap Loads** button in the phone load to make it the default phone load.

Step 4 Click **Save**.

Configure Self-Provisioning Authentication

The Phone Migration Service uses Self-Provisioning system settings to authenticate users prior to phone migration.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **Self-Provisioning**.

Step 2 Configure whether you want the self-provisioning to authenticate end users by clicking one of the following radio buttons:

- **Require Authentication**—In order to use self-provisioning, end users must enter their password, PIN, or a system authentication code.
- **No Authentication Required**—End users can access the self-provisioning options without authenticating.

Step 3 If the self-provisioning functionality is configured to require authentication, click one of the following radio buttons to configure the method whereby the self-provisioning options authenticates end users:

- **Allow authentication for end users only**—End users must enter their password or PIN.
- **Allow authentication for users (via Password/PIN) and Administrators (via Authentication Code)**—End Users must enter an authentication code. If you choose this option, configure the authentication code by entering an integer between 0 and 20 digits in the **Authentication Code** text box.

Step 4 Click **Save**.

Phone Migration Tasks

After self-provisioning Authentication is setup, use any of the following procedures to migrate phones.

Migrate Phones Using Phone Migration Service (Administrator)

Administrators can use this procedure to migrate Cisco IP Phones on behalf of an end user, or to migrate common phones (for example, a lobby phone).

Before you begin

Make sure that the old phone is in the "Unregistered" state before you proceed with migration. You can plug the new phone into the network, wait until you get the phone migration or provisioning prompt and start the migration process. Once the migration is successful, the device re-registers with the users phone configuration data.

Procedure

Step 1 Connect the new Cisco IP Phone to the network.

Step 2 Select option **2** to replace an existing phone.

Note
If Administrator has not configured Activation Code based device onboarding or phone migration is done on the 11.5(1)SU8 version of Unified Communications Manager which does not support activation code, the phone displays the screen to enter the primary extension.

Step 3 Enter the primary extension number of your phone.

Step 4 Enter the **Authentication Code**.

Migration starts after a successful authentication. After successful migration, the phone restarts with configuration settings migrated from your old phone.

Note
Consider a scenario where the **On-premise Onboarding Method** option is selected as **Auto registration** in the Device Defaults Configuration page. While performing the phone migration, if you press either the **EXIT** or **BACK** button when an error message is displayed, it takes you directly to the initial phone migration screen after a possible expected delay. This delay is due to the phones trying to re-register to the Unified CM as auto-registration is disabled in the Unified Communications Manager server.

Note
If there are more than one device for the user with the same primary extension, user will be prompted to choose the device to be migrated. For more information, see Phone Migration Service—User Assigned with Multiple Devices, on page 716.

Migrate Phones Using Phone Migration Service (Phone Users)

Phone users can use this procedure to migrate to a new Cisco IP Phone using a non-IVR method. After you plug in the phones to the network, the phone will attempt to boot and configure itself. From the default phone load, users get options to select either the 'Provision a new phone' or 'Replace an existing phone'.

Note An end user should perform migration only if they are the owner of the old device.

Note If you decide to choose the **Retain Existing Phone(s)**' option from the **When Provisioning a Replacement Phone for an End User** enterprise parameter and you migrate the phone using the Phone Template, the old migrated phone configurations details will not be listed in the Find List page. Only the successful new phone migration details are listed. This is because the **Retain Existing Phone(s)** option is applicable only for Phone Migrations based on Phone Types and not based on the Phone Template.

Before you begin

Make sure that the old phone is in the "Unregistered" state before you proceed with migration. You can plug the new phone into the network, wait until you get the phone migration or provisioning prompt and start the migration process. Once the migration is successful, the device re-registers with the users phone configuration data.

Procedure

Step 1 Connect the new Cisco IP Phone to the network.

Step 2 Select the option **2** to replace an existing phone.

Note
If the Administrator hasn't configured Activation Code based device onboarding or phone migration is done on the 11.5(1)SU8 version of Unified Communications Manager which doesn't support activation code, the phone displays the screen to enter the primary extension.

Step 3 Enter the primary extension number of your phone.

Step 4 Enter your **PIN**.

Migration starts after a successful authentication. After successful migration, the phone restarts with configuration settings migrated from your old phone.

Note
Consider a scenario where the **On-premise Onboarding Method** option is selected as **Auto registration** in the Device Defaults Configuration page. While performing the phone migration, if you press either the **EXIT** or **BACK** button when an error message is displayed, it takes you directly to the initial phone migration screen after a possible expected delay. This delay is due to the phones trying to re-register to the Unified CM as auto-registration is disabled in the Unified Communications Manager server.

Note
If there are more than one device for the user with the same primary extension, the user will be prompted to choose the device to be migrated. For more information, see Phone Migration Service—User Assigned with Multiple Devices, on page 716.

Phone Migration Service COP File

If you are running any version of the Unified Communications Manager starting from 11.5(1) until 11.5(1)SU7, install Phone Migration Service COP file (ciscocm-migration-service-11-5-1.zip) to get the Native Phone Migration feature support.

As part of the COP file installation, ‘Tftp restart’ service is done automatically for Unified Communications Manager.



Note If you plan to upgrade the Unified CM after Phone Migration Service COP file installation, ensure that you upgrade your Unified CM server to a release version that has native support for the Native Phone Migration feature.

View Phone Migration Report

Use this procedure to view the list of all the Cisco IP Phones that are migrated.

Procedure

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

Step 2 From the Find and List Phones page, choose **Migrate (old phone)** from the **Find Phone** where drop-down list.

Step 3 Click **Find**.

You can view the list of all the old devices that got migrated. This list is populated only if the **Retain Existing Phone** option is configured in the Enterprise Parameters page.

Migrate Phones using Cisco Unified CM Administration Interface

Use this procedure to migrate phones using either the **Phone Template** or **Phone Type (and Protocol)** options in the Cisco Unified CM Administration interface.

Procedure

Step 1 In the Find and List Phones window (**Device > Phone**), find the Cisco IP Phone that you want to migrate.

Step 2 In the Phone Configuration window for the Cisco IP Phone that you want to migrate, choose **Migrate Phone** from the **Related Links** drop-down list.

Step 3 To migrate phones, you can use one of the following options:

a) **Phone Template**—Choose the phone template for the phone model to which you want to migrate the phone configuration.

b) **Phone Type (and Protocol)**—Choose the Cisco IP Phone model for which you want to migrate the phone configuration.

Step 4 Enter the **MAC address** for the new Cisco Unified IP Phone to which you are migrating the configuration.

Step 5 **(Optional)** Enter a description for the new phone. For more information on the migration considerations and configuration settings, see the *Cisco Unified CM Administration Online Help* pages.

Step 6 Click **Save**.

If a warning message displays that the new phone may lose feature functionality, click **OK**. After migration, the new device will inherit setting of the old phone.

Migration Scenarios

Phones Using Shared Lines

Consider a scenario where an old phone has the primary DN shared line with multiple devices. These devices may be owned by the same user or among multiple users. When you try to migrate the old phone with a new Cisco IP Phone using Self-Provisioning IVR or Phone Migration Service methods using a shared line, migration is possible only for users who own the device with the DN as Line 1. Here, the shared lines feature settings are carried over after phone migration.

If the old phone does not support the Shared Line feature, the old phones lines are removed after phone migration. The new phone retains the old phones lines after phone migration.

Phone Migration Service Running on Proxy TFTP

Native phone migration using Phone Migration Service support Cisco proxy TFTP server deployment models.

Phone Migration Service running on the proxy TFTP searches for the primary extension on all the remote clusters and redirect the phone to its home or local Phone Migration Service for completing phone migration.

In a proxy setup environment, Unified Communications Manager server looks for devices with same DN in a remote cluster. If there is more than one device in the registered or unregistered state with the same DN in any one of the off-cluster, those devices from that particular off-cluster is not considered for phone migration. This is a known limitation for Release 11.5(1)SU8.

Note In Proxy TFTP, after migration, the phone restarts with configuration settings migrated from your old phone. Note that the phone takes 2 reset cycles to restart after successful migration.

Phone Migration Service—User Assigned with Multiple Devices

If there is more than one device for the user with the same primary extension for migration, user will be prompted to choose the device to be migrated.

The following table displays the various migration scenarios possible:

Table 68: Device Listing and Migration Scenarios

	Device Status Before Phone Migration	Phone Display During Migration
Scenario 1	Device 1—Registered Device 2—Unregistered	Phone Configuration settings of Device 2 will be migrated.
Scenario 2	Device 1—Registered Device 2—Registered Device 3—Unregistered	Phone Configuration settings of Device 3 will be migrated.

	Device Status Before Phone Migration	Phone Display During Migration
Scenario 3	Device 1—Registered Device 2—Unregistered Device 3—Unregistered	Displays the device list with the following: Description, Phone Model, and MAC Address. Users should choose the device from the list required for phone migration.
Scenario 4	Device 1—Unregistered Device 2—Unregistered Device 3—Unregistered	Displays the device list with the following: Description, Phone Model, and MAC Address. Users should choose the device from the list required for phone migration.
Scenario 5	Device 1—Registered Device 2—Registered Device 3—Registered	Displays the device list with the following: Description, Phone Model, and MAC Address. Users should choose the device from the list required for phone migration.
Scenario 6	More than 3 devices are in Registered or Unregistered states	Displays the device list with the following: Description, Phone Model, and MAC Address. Users should choose the device from the list required for phone migration.

Device Display Based on Unified CM Parameter Settings

The following table lists the behavior of the migration screen on the new phone based on the release version of Cisco Unified Communications Manager and related configuration settings:

Table 69: Device Display Based on Various Unified CM Parameter Settings

Phone or Device preconfigured by Administrator	Autoregistration Enabled at Enterprise Level	Onboarding Method at Device Defaults Level	Behavior without Phone Migration Service	Behavior with Phone Migration Service
No	Yes	—	Devices get auto registered to the network.	Devices get auto registered to the network.
No	No	—	Device retries to register to the network and keeps retrying based on the back-off timer set.	Phone replacement screen prompts to enter the Primary Extension and PIN.
Yes	N/A	—	Device registers with the preconfigured settings.	Device registers with the preconfigured settings.

Phone or Device preconfigured by Administrator	Autoregistration Enabled at Enterprise Level	Onboarding Method at Device Defaults Level	Behavior without Phone Migration Service	Behavior with Phone Migration Service
No	Yes	Device Type set to Activation Code	Welcome screen of the phone prompts the following: "Enter activation code".	Phone replacement screen prompts to select one of the following: “Provision a new phone” or “Replace an existing phone”.
No	No	Device Type set to Auto Registration	Device retries to register to the network and gives up.	Phone replacement screen prompts the following: “Replace an existing phone”.
No	No	Device Type set to Activation Code	Welcome screen of the phone prompts the following: "Enter activation code".	Phone replacement screen prompts to select one of the following: “Provision a new phone” or “Replace an existing phone”.
Yes	N/A	N/A	Device retries to register to the network and keeps retrying based on the back-off timer set.	Device registers with the preconfigured settings.

Phones Using Extension Mobility

In scenarios where the old phone supports Extension Mobility login, after migration the new device will also support the Extension Mobility capability. If the old phone was logged-in before migration, then the logged-in Extension Mobility user is automatically logged-out during the phone migration. The user must perform fresh Extension Mobility login on the new phone.

Note Native phone migration does not support end-users Extension Mobility Device profile migration.

CTI Controlled Devices

If an old device is CTI Controlled before phone migration, then the new device is also CTI controlled. This is because the device configuration settings are carried over after phone migration.

Phones with Key Expansion Module

If the Key Expansion Module (KEM) attached to your old phone is not compatible with your new phone model, you will lose the KEM 'Expansion Module Information' configuration settings on the new phone after phone migration.

The following table explains different scenarios:

Table 70: KEM Migration Scenarios

Scenarios	Old Phone (Model 79xx)	New Phone (Model 88xx)	Expected Behavior After Migration
• KEM 1 attached to the old phone is compatible with the new phone. • User removes the KEM 1 from the old phone and attaches it with the new phone.	KEM 1	KEM 1	KEM 1 configuration settings are carried over.
• KEM 1 attached to the old phone is incompatible with the new phone. • User attaches a compatible KEM 2 (brand new or used) to the new phone.	KEM 1	KEM 2	KEM 1 configuration settings are carried over to KEM 2.
• KEM 1 attached to the old phone is deprecated. • User attaches a new KEM 3 to the new phone.	KEM 1	KEM 3	KEM 1 configuration settings are carried over to KEM 3.

Product Specific Configuration Parameters

During phone migration, the product-specific parameters of the older phones are also migrated. The new phone considers only the parameters that the phone understands. The remaining parameters are set to their default value.

If the ‘Line Mode’ parameter is already configured during phone migration, the ‘Line Mode’ configuration settings are carried over. Else, this parameter is set as ‘Session Line Mode’ by default.

Also, during phone migration, if the "Show All Calls on Primary Line" parameter is configured in the old phone, then the new phone will retain the "Show All Calls on Primary Line" parameter after phone migration. If this parameter is not configured before phone migration, it is enabled by default after phone migration.

Phone Button Templates

When a SCCP Phone model is migrated to a SIP Phone model, the SIP Phone considers only the parameters that it understands. Else, it takes the default configuration values from the SIP Phone model (for example, Standard SIP Profile).

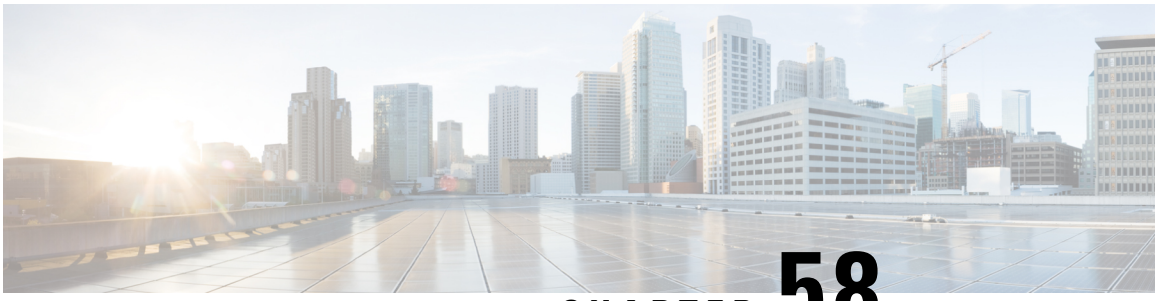
If the old phone had a unique Custom Phone Button Template, then the new phone will retain the Custom Phone Button Template after phone migration. For the Standard Phone Button template, the new device uses the Standard Phone Button Template specific to the phone model.

Table 71: Phone Button Templates-Migration Scenarios

Old Device	Phone Button Template of the Old Device	New Device Selected for Phone Migration	Phone Button Template of the New Device After Phone Migration
Cisco Unified IP Phone 7965 SCCP	Standard 7965 SCCP	Cisco IP Phone 8861	Standard 8861 SIP
Cisco Unified IP Phone 7965 SCCP	Universal Device Template Button layout	Cisco IP Phone 8861	Universal Device Template Button layout
Cisco Unified IP Phone 7965 SCCP	Custom 7965 SCCP	Cisco IP Phone 8861	Custom 7965 SCCP
Cisco Unified IP Phone 8851 SIP	Standard 8851 SIP	Cisco IP Phone 8861	Standard 8861 SIP

Collaboration Devices—Room Systems, Desk, and IP Phones

- You can only migrate a video endpoints device to another video endpoints device.
- Unified Communications Manager does not support phone migration to CSF (Jabber for Desktop), TCT (Jabber for iPhone), TAB (Jabber for iPad), or BOT (Jabber for Android) devices.
- You cannot migrate video endpoints using Phone Migration Service.
- If the old video endpoints device was in the locked state, note that the new video endpoints device will not retain the locked state after phone migration.



CHAPTER 58

Video Endpoints Management

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- Video Endpoints Management Feature Compatibility, on page 722
- Migration Considerations for Video Endpoints Provisioning, on page 723
- Video Endpoints Migration Report, on page 724
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Video Endpoints Management Overview

This feature simplifies the administrator's job of provisioning and managing Cisco TelePresence video endpoints. An administrator can provision settings for Cisco TelePresence endpoints in Unified Communications Manager and then push those Product-Specific Configuration settings to endpoints.

Prior to Release 12.5(1)SU1, only a limited set of Product-Specific Configurations were pushed from Unified Communications Manager to the endpoint resulting in a partial configuration of the endpoint. Administrator had to rely on Cisco TelePresence Management Suite or TelePresence Endpoint's web interface to configure all the settings. The Phone Configuration window in Unified Communications Manager contains a complete Product-Specific Configuration layout for Cisco TelePresence endpoints that matches what users see on their endpoint. This update lets administrators apply settings on behalf of users and then push those settings to users.

Note The Bulk Administration Tool (BAT) **Phone Template Configuration** page also displays the new model-specific configurations in a tabbed layout, supporting the complete list of endpoint parameters. You can import the entire set of parameters or modify a specific parameter in the endpoint in bulk.

Video endpoints managements feature provides the following benefits:

- TelePresence endpoints can be fully provisioned from Unified Communications Manager—Endpoints parameters listed in the Unified Communications Manager user interface are in the same order as listed in the **Advanced Configuration** settings of your Cisco TelePresence model. For more information on the various advanced parameters, see the respective model in the Collaboration Endpoints Administrator Guides.
- New **Product-Specific Configuration** Layout—New layout details the model-specific configurations in a tabbed layout. This is an upgrade from the earlier flat format that provided access only to a limited

set of parameters. The new layout ensures that you have a complete list of Cisco TelePresence settings on the Cisco Unified CM Administration interface.

- Automatic migration of the configuration data from the video endpoints—This simplifies the deployment of endpoints by automatically synching data from endpoints to Unified Communications Manager and vice versa. Endpoint configurations can be fully restored in case of reset to factory settings or Product Returns & Replacements (RMA) swaps.

Note Any endpoint that supports Collaboration Endpoint (CE) Software 9.8 or higher can use this new provisioning layout for the Product-Specific Configuration fields on the Phone Configuration page. If you are using a CE software version prior to 9.8, you will be able to view all the new set of advanced parameters; but, the new set of parameters functions only if you upgrade your CE Software version to 9.8 or higher. The subset of parameters supported is marked with a "#" to the right of each parameter value in the user interface. You must load a device pack onto Unified Communications Manager if a device type is capable of supporting the new provisioning framework, but does not show the additional parameters.

Video Endpoints Management Feature Compatibility

Following table details the video endpoints management feature compatibility with Unified Communications Manager and Collaboration Endpoint (CE) versions:

Unified Communications Manager Version	CE Endpoint Version	Expected Behavior
12.5(1) SU1	9.8 and above	Devices added prior to 12.5(1) SU1: • Advanced Configuration UI (Tabbed Layout) for successfully backed up devices • Limited Configuration UI (Flat Layout) for devices yet to be backed up New device added through UI/BAT/AXL: • Advanced Configuration UI **Note** It's highly recommended that you run CE 9.8 or higher.

Unified Communications Manager Version	CE Endpoint Version	Expected Behavior
12.5(1) SU1	9.7 and below	Devices added prior to 12.5(1) SU1: • Limited Configuration UI New device added through UI/BAT/AXL: • Advanced Configuration UI with only a limited set of parameters taking into effect **Note** For migrations with CE 9.7 or earlier, you cannot maintain the existing endpoint configuration during migration. When the migrated device registers, Unified CM overwrites the existing configuration with default settings.
12.5(1) and below	9.8 and above	Limited Configuration UI

Migration Considerations for Video Endpoints Provisioning

Auto Backup After Unified Communications Manager Upgrade

When upgrading to Unified Communications Manager 12.5(1)SU1, the existing configuration data for the supported endpoint types is automatically migrated from endpoints to Unified Communications Manager.

1. Upgrade Unified Communications Manager to version 12.5(1)SU1 or later.
2. Endpoints register to Unified Communications Manager.
3. Unified Communications Manager then sends a SIP Notify message to endpoints requesting for the full set of Product-Specific Configuration parameters.
4. Endpoints that are upgraded to CE 9.8 and above send full set of configuration data to Unified Communications Manager (in xConfiguration format) using a SIP REFER message.
5. Unified Communications Manager processes this configuration data and populates the complete list of Cisco TelePresence settings (Advanced Configuration UI) on the Cisco Unified CM Administration interface.



Note Unified Communications Manager server displays the complete endpoint configuration settings in the new layout only if Unified CM is able to successfully back up data from the endpoint.

Configuration Control Modes

Based on the deployment needs, administrators can configure various configuration control modes in the Cisco Unified CM Administration interface. You can decide whether you want to control the configuration settings centrally from the endpoints or Unified Communications Manager or both of them together.

Navigate to the Product-Specific Configuration Layout section on the Phone Configuration page and choose the **Configuration Control Mode** under "General Settings" in the Miscellaneous tab to control the various modes. Following are the various Configuration Control Modes:

- **Unified CM and Endpoint (Default)**—Use this mode if you want Unified Communications Manager and endpoint to operate as the multi-prime source for provisioning endpoint data. If Unified CM and Endpoint is the configured mode, any update made via an endpoint locally is synched with the Unified CM server.
- **Unified CM**—Use this mode if you want Unified Communications Manager to operate as the centralized primary source for provisioning endpoint data and does not want to accept any configurations done from the endpoints locally.
- **Endpoint**—Use this mode if you want endpoints to operate as the centralized primary source of configuration data. In this mode, endpoint ignores any configuration data from Unified Communications Manager and doesn't sync back the changes done locally. This mode is typically used when an Audiovisual (AV) integrator is installing the endpoints and wants to control configuration from the endpoint.



Note In the **Endpoint** mode, CE devices continue to accept that limited set of parameters supported prior to release 12.5(1)SU1. Unified Communications Manager indicates these parameters with a "#" symbol. CE devices will ignore the extended set of parameters supported from the 12.5(1)SU1 release onwards.

On-demand Configuration Pull Functionality

Administrators can use the **Get Config from Phone** option to pull configuration changes from the CE 9.8 endpoint devices on-demand at that given point.

Navigate to the Product-Specific Configuration Layout section on the Phone Configuration page and click the **Get Config from Phone** button on the top corner of the page to pull any data configuration from the CE 9.8 endpoints on-demand. This option is enabled only if the endpoint is in the registered state.

Video Endpoints Migration Report

Video Endpoint with Extended Configuration Backup is the new filter is introduced on the Find and List Phones window for release 12.5(1)SU1. Administrators can search for details on how many CE endpoints got migrated automatically and how many CE endpoints did not. Based on this information, they can take corrective measures.



Note In the Find and List Phones window, the **Video Endpoint with Extended Configuration Backup** filter is applicable only for video endpoints running Collaboration Endpoint (CE) Software 9.8 or higher.

Provisioning and Migration Scenarios

The following table describe various provisioning and migration scenarios. All of these scenarios assume that your TelePresence video endpoints are upgraded to a CE release that supports Product-Specific Configuration provisioning from Unified CM. In Unified CM, these settings appear in the **Product-Specific Configuration** section, but on the endpoint, they appear under **Advanced Configuration**.

Table 72: Provisioning and Migration Scenarios for Video Endpoints

Task	Existing Configuration Summary	What to do
Provisioning New Video Endpoints	• Brand new device • Device is not provisioned on Unified CM • No existing settings on the device or on Unified CM	With Unified CM at a minimum release 12.5(1)SU1 and the CE endpoint at 9.8, you can provision new endpoints and manage the product-specific configurations from Unified CM.
Migrating Existing Video Endpoints from VCS	• Existing device • Device is not provisioned on Unified CM • Device is configured, but Unified CM does not have any of the configurations	If you are migrating existing video endpoints from a Cisco TelePresence Video Communications Server to Cisco Unified Communications Manager: **Adding Phones via Phone Configuration window in Unified CM:** • Add the phone to Unified CM, but DO NOT CLICK **Save**. • Register the phone. After registration, the existing **Advanced Configuration** settings from the phone are uploaded to Unified CM and display in **Product-Specific Configurations** in the **Phone Configuration** window. • In the **Phone Configuration** window, configure the new settings and click **Save**. The provisioned settings download to the phone. For a detailed procedure, see Add Migrating Video Endpoint to Unified CM, on page 726 **Adding Phones via Bulk Administration** Make sure that the csv file or BAT Template that you use for provisioning does not include the Product-Specific Configuration fields. **Adding Phones via AXL** Make sure that the AXL request does not include any Product-Specific Configuration fields.

Task	Existing Configuration Summary	What to do
Upgrading from an Earlier Release of Unified CM with Registered Video Endpoints	• Existing device • Device is provisioned on a pre-12.5 release of Unified CM • Unified CM has a limited set of Product-Specific Configuration settings for the device	So long as the CE endpoint is at a supported version, when you upgrade Unified CM, the **Advanced Configuration** settings from the endpoint get pulled into Unified CM automatically following device registration and display under the **Product-Specific Configuration** section of the **Phone Configuration** window. After registration, you can set the **Configuration Control Mode** in addition to whatever settings you want.

Add Migrating Video Endpoint to Unified CM

If you are migrating existing Cisco TelePresence Video Endpoints from a Cisco TelePresence Video Communications Server to Unified Communications Manager, use this procedure to add the CE endpoint into Unified CM via the **Phone Configuration** window so that the existing **Advanced Configurations** from the endpoint can be managed from the **Phone Configuration** window in Unified CM.

Note Make sure to follow this procedure closely. The settings from the endpoint do not automatically upload to Unified CM until after device registration.

Note This procedure uses the **Add New from Template** setting in the Unified CM **Phone Configuration** window. You can also use tools like Bulk Administration or AXL to add the endpoint.

Before you begin

It's highly recommended that you upgrade firmware to CE 9.8 or higher before you migrate. With CE 9.7 or earlier, Unified CM overwrites the existing endpoint configuration during registration with default settings.

Procedure

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

Step 2 Click **Add New from Template** and enter the following phone details:

- Select the model from the **Phone Type** drop-down list.
- Enter the **MAC Address** of the endpoint
- From the **Device Template**, select a Universal Device Template.
- Select the **Directory Number** that you want to add to the phone. If none exists, click **New** and configure a directory number.
- From the **User** drop-down list, select the user whom will own the device.

Step 3 Click **Add**. The **Phone Configuration** displays with the universal device template settings filling out the phone configuration. The **Product-Specific Configuration** section also appears, but with default settings, rather than the existing settings from the phone.

Note
You can also add the device using the **Phone Configuration** window's **Add New**, but this method requires that you enter settings manually.

Step 4 DO NOT CLICK **Save**. If you save settings, Unified CM does not load existing settings from the phone. If you saved by mistake, go straight to the troubleshooting Note at the bottom of this procedure for recovery steps.

Step 5 Register the phone.
During registration, the existing **Advanced Configuration** settings from the phone get pulled into Unified CM and display in the **Phone Configuration** window's **Product-Specific Configuration** section.

Step 6 In the **Phone Configuration** window, configure how you want endpoint settings to be managed by configuring the **Configuration Control Mode** field:

- **Unified CM and Endpoint (Default)**—Use this mode if you want Unified Communications Manager and endpoint to operate as the multi-prime source for provisioning endpoint data. If Unified CM and Endpoint is the configured mode, any update made via an endpoint locally is synched with the Unified CM , and any change made on Unified CM syncs to the endpoint.
- **Unified CM**—Use this mode if you want Unified Communications Manager to operate as the centralized primary source for provisioning endpoint data and does not want to accept any configurations done from the endpoints locally.
- **Endpoint**—Use this mode if you want endpoints to operate as the centralized primary source of configuration data. In this mode, the endpoint maintains existing settings, ignores any configuration data from Unified Communications Manager, and doesn’t sync back the changes done locally. This mode is typically used when an Audiovisual (AV) integrator is installing the endpoints and wants to control configuration from the endpoint.

Note
If you want to maintain existing settings on the endpoint, it's recommended to choose **Endpoint** mode, at least until after the endpoint has completed the registration process in full. You can switch the configuration to one of the other modes after you complete this procedure.

Step 7 Configure any phone settings that you want. For more information on the fields and their configuration options, see Online Help.

Step 8 Click **Save**.
The provisioned settings in Unified Communications Manager download to the endpoint.

Note If you clicked **Save** mistakenly in the **Phone Configuration** window prior to device registration, the existing **Advanced Configuration** settings from the endpoint will not load to Unified CM when the device registers. To recover, do the following prior to device registration:

- In Unified CM, set the **Configuration Control Mode** to **Endpoint** and click **Save**.
- Let the phone register to Unified CM.
- After registration, return to the device configuration in the **Phone Configuration** window and click the **Get Config from Device** button. The setting results in the existing **Advanced Configurations** on the phone getting pulled into Unified CM. Note that this button does not appear until after device registration.
- Return to Step 6 of the procedure in order to complete the configuration.



PART XIV

Advanced Call Processing

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CHAPTER 59

Configure External Call Control

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External Call Control Overview

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

The adjunct router influences how your system allows, diverts, or denies calls; modifies calling and called party information; plays announcements to callers; resets call history so that adjunct voicemail and IVR servers can properly interpret calling and called party information; and logs reason codes that indicate why calls were diverted or denied.

External call control provides the following functions:

- Best Quality Voice Routing—The adjunct route server monitors network link availability, bandwidth usage, latency, jitter, and MOS scores to ensure that calls are routed through voice gateways that deliver the best voice quality to all call participants.
- Least Cost Routing—The adjunct route server is configured with carrier contract information such as local access and transport area (LATA) and inter-LATA rate plans, trunking costs, and burst utilization costs to ensure that calls are routed over the most cost effective links.
- Ethical Wall—The adjunct route server is configured with corporate policies that determine reachability, for example, whether user 1 is allowed to call user 2.

External Call Control Prerequisites

This feature requires the Cisco Unified Routing Rules XML Interface, which directs your system on how to handle calls.

For more information, see the *Cisco Unified Routing Rules Interface Developers Guide* (CURRI documentation) at https://developer.cisco.com.

External Call Control Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Configure a Calling Search Space for External Call Control, on page 733	Configure a calling search space for your system to use when the route server sends a divert obligation. A calling search space comprises an ordered list of route partitions that you assign to devices. Calling search spaces determine the partitions that calling devices search when they attempt to complete a call.
Step 2	Configure an External Call Control Profile, on page 734	Configure an external call control profile to provide the URIs for the adjunct route server, a calling search space that is used for diverting calls, a timer that indicates how long your system waits for a response from the adjunct route server, and so on.
Step 3	Assign a Profile to a Translation Pattern, on page 734	For the translated patterns that you want to use with external call control, assign an external call control profile to the pattern. When a call occurs that matches the translation pattern, your system immediately sends a call routing query to an adjunct route server, and the adjunct route server directs your system on how to handle the call.
Step 4	(Optional) Import the Route Server Certificate into the Trusted Store, on page 735	If the route server uses HTTPS, import the certificate for the route server into the trusted store on your system node. You must perform this task on each node in the cluster that can send routing queries to the route server. If you use HTTPS for the primary or secondary web service URIs in the external call control profile, your system uses certificates to mutually authenticate through a TLS connection to the configured adjunct route servers.

	Command or Action	Purpose
Step 5	(Optional) Export the Self-Signed Certificate to the Route Server, on page 735	If the route server uses HTTPS, export the Cisco Unified Communications Manager self-signed certificate to the route server. You must perform this task for each node in the cluster that can send routing queries to the route server. To ensure that the primary and redundant route servers can authenticate with Cisco Unified Communications Manager through HTTPS, you must generate a self-signed certificate that you can import to each adjunct route server that sends directives to your system. Perform this procedure for each node in the cluster that can contact the primary and redundant adjunct route server.
Step 6	(Optional) Configure the Chaperone Function, on page 736	Configure chaperone functionality if your routing rules from the route server state that a chaperone must monitor or record a call. A chaperone is a designated phone user who can announce company policies in the call, monitor the call, and record the call.
Step 7	(Optional) Configure Customized Announcements, on page 737	Follow this procedure if your routing rules require that an announcement is played for some calls and you do not want to use the Cisco-provided announcements.

Configure a Calling Search Space for External Call Control

Configure a calling search space for your system to use when the route server sends a divert obligation. A calling search space comprises an ordered list of route partitions that you assign to devices. Calling search spaces determine the partitions that calling devices search when they attempt to complete a call.

Procedure

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

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

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

Step 4 In the **Description** field, enter a description.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

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

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

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

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

Step 8 Click **Save**.

Configure an External Call Control Profile

Configure an external call control profile to provide the URIs for the adjunct route server, a calling search space that is used for diverting calls, a timer that indicates how long your system waits for a response from the adjunct route server, and so on.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **External Call Control Profile**.

Step 2 Perform one of the following tasks:

- Click **Find** and then choose an existing external call control profile from the resulting list to modify the settings for an an existing external call control profile, enter search criteria.
- Click **Add New** to add a new external call control profile.

Step 3 Configure the fields on the **External Call Control Profile Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 4 Click **Save**.

Assign a Profile to a Translation Pattern

Configure an external call control profile to provide the URIs for the adjunct route server, a calling search space that is used for diverting calls, a timer that indicates how long your system waits for a response from the adjunct route server, and so on.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Translation Pattern**.

Step 2 Perform one of the following tasks:

- Click **Find** and then choose an existing translated pattern from the resulting list to modify the settings for an existing translated pattern, enter search criteria, .
- Click **Add New** to add a new translated pattern.

Step 3 From the **External Call Control Profile** drop-down list, choose the external call control profile that you want to assign to the pattern.

Step 4 Configure other fields as needed in the **Translation Pattern Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 5 Click **Save**.

Import the Route Server Certificate into the Trusted Store

If the route server uses HTTPS, import the certificate for the route server into the trusted store on your system node. You must perform this task on each node in the cluster that can send routing queries to the route server. If you use HTTPS for the primary or secondary web service URIs in the external call control profile, your system uses certificates to mutually authenticate through a TLS connection to the configured adjunct route servers.

Procedure

Step 1 From Cisco Unified Operating System Administration, choose **Security** > **Certificate Management**.

Step 2 Click **Upload Certificate**.

Step 3 In the **Upload Certificate** popup window, click **CallManager-trust** from the **Certificate Name** drop-down list, and browse to the certificate for the adjunct route server.

Step 4 After the certificate appears in the **Upload File** field, click **Upload**.

Step 5 (Optional) Perform this procedure again if your system can contact a redundant adjunct route server.

Export the Self-Signed Certificate to the Route Server

If the route server uses HTTPS, export the Unified Communications Manager self-signed certificate to the route server. You must perform this task for each node in the cluster that can send routing queries to the route server. To ensure that the primary and redundant route servers can authenticate with Unified Communications Manager through HTTPS, you must generate a self-signed certificate that you can import to each adjunct route server that sends directives to your system.

Perform this procedure for each node in the cluster that can contact the primary and redundant adjunct route server.

Procedure

Step 1 From Cisco Unified Operating Administration, choose **Security** > **Certificate Management**.

Step 2 In the **Certificate List** window, click **Generate New**.

Step 3 From the **Certificate Name** drop-down list, choose **CallManager**.

Step 4 Click **Generate New**.

Step 5 From the **Find and List Certificates** window, choose the **CallManager.pem** certificate that you just created.

Step 6 After the certificate file data appears, click **Download** to download the certificate to a location that you can use for exporting the certificate to the adjunct route server.

Step 7 Export the certificate to each adjunct route server that sends directives.

Configure the Chaperone Function

Configure chaperone functionality if your routing rules from the route server state that a chaperone must monitor or record a call. A chaperone is a designated phone user who can announce company policies in the call, monitor the call, and record the call.

Unified Communications Manager provides the following capabilities to support chaperone functionality, as directed by the adjunct route server:

- Redirect an incoming call to a chaperone, hunt group, or a list of chaperones.
- Provide a chaperone with the ability to record a call.

When the chaperone is connected to the caller or when the chaperoned conference is established, the **Record** softkey or programmable line key (PLK) (depending on the phone model) is active on the phone so that the chaperone can invoke call recording. Call recording occurs for only the current call, and call recording stops when the current call ends. Messages that indicate the status of recording may display on the phone when the chaperone presses the recording softkey or PLK.

Procedure

Step 1 For phones on which you want to enable recording, set the Built-in-Bridge to **On** in the **Phone Configuration** window.

Step 2 Create a recording profile:

a) Choose **Device** > **Device Settings** > **Recording Profile**.
b) Create a Call Recording Profile for the phones that can record chaperoned conferences.

Step 3 Apply the recording profile to the line appearance.

Step 4 Add a SIP trunk to point to the recorder.

Step 5 Create a route pattern that points to the SIP trunk.

Step 6 Configure the following service parameters:

a) Play Recording Notification Tone to Observed Target
b) Play Recording Notification Tone to Observed Connected Target

Step 7 Assign the Standard Chaperone Phone softkey template to the phone that the chaperone uses.

Step 8 Perform the following steps from **Call Routing** > **Directory Number** for a new phone or from **Device** > **Phone** if the phone is already configured:

a) Configure only one directory number (DN) for the chaperone phone.
b) For the DN on the chaperone phone, choose **Device Invoked Call Recording Enabled** from the **Recording Option** drop-down list.
c) For the DN on the chaperone phone, enter `2` for the **Maximum Number of Calls** setting, and enter `1` for the **Busy Trigger** setting.

Step 9 For Cisco Unified IP Phones that support the **Record** softkey, configure the Standard Chaperone Phone softkey template so that only the **Conference**, **Record**, and **End Call** softkeys display on the phone in a connected state.

Step 10 For Cisco Unified IP Phones that support the record programmable line key (PLK), configure the PLK in the **Phone Button Template Configuration** window.

Step 11 (Optional) If you have more than one chaperone in your cluster, add the chaperone DN to the chaperone line group that you plan to assign to the chaperone hunt list.

This step ensures that an available chaperone monitors the call.

Configure Customized Announcements

Follow this procedure if your routing rules require that an announcement is played for some calls and you do not want to use the Cisco-provided announcements.



Tip Do not use embedded spaces for the announcement identifier.

If other language locales are installed, you can upload other .wav files for this announcement to use with those locales.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Media Resources** > **Announcement**.

Step 2 Perform one of the following tasks:

- To add a new announcement:

a) Click **Add New**.
b) In the **Announcement Identifier** field, enter an announcement identifier.
c) In the **Description**, enter a description of the announcement.
d) From the **Default Announcement** drop-down list, choose a default Cisco-provided announcement if desired.
e) Click **Save**.

- To upload a custom .wav file for the announcement:

a) Click **Upload File**.
b) From the **Locale** drop-down list, choose the locale language for the announcement.
c) Click **Choose File**, and then choose a .wav file to upload.
d) Click **Upload File**.
e) When the upload finishes, click **Close** to refresh the window and show the uploaded announcement.

External Call Control Interactions

Table 73: External Call Control Interactions

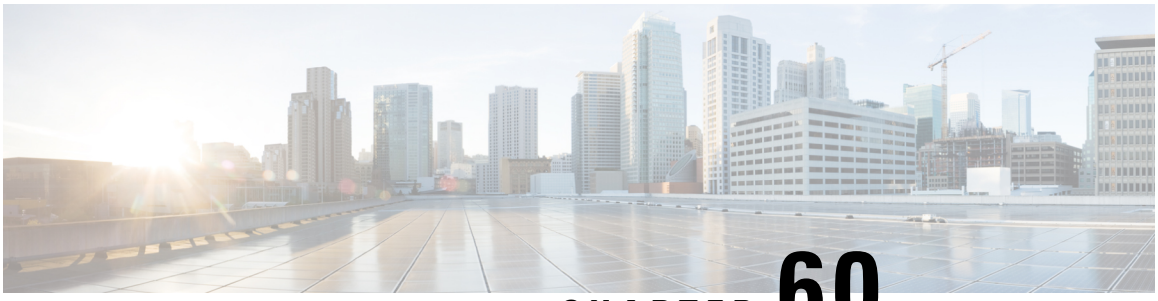
Feature	Interaction
Best Call Quality Routing	You can set up routing rules on the adjunct route server that determine which gateway to use for a call, taking voice quality into consideration. For example, gateway A provides the best voice quality, so it is used for the call. In this case, the adjunct route server monitors network link availability, bandwidth usage, latency, jitter, and mean opinion scores (MOS) to ensure that calls are routed through voice gateways that deliver the best voice quality to all call participants.
Call Detail Records	External Call Control functions can be displayed in call detail records; for example, the call detail record can indicate whether the adjunct route server permitted or rejected a call. In addition, the call detail record can indicate whether Unified Communications Manager blocked or allowed calls during which it did not receive a decision from the adjunct route server.
Call Forward	External Call Control intercepts calls at the translation pattern level, while Call Forward intercepts calls at the directory number level. External Call Control has a higher priority than Call Forward; for calls that invoke Call Forward, Unified Communications Manager sends a routing query to the adjunct route server if the translation pattern is assigned to an External Call Control profile. Call Forward is triggered only when the adjunct route server sends a Permit decision with a Continue obligation to the Cisco Unified Communications Manager. **Note** The **Call Diversion Hop Count** service parameter that supports External Call Control and the **Call Forward Call Hop Count** service parameter that supports Call Forward are independent of each other.
Call Pickup	When a phone user tries to pick up a call by using the Call Pickup feature, External Call Control is not invoked; Unified Communications Manager does not send a routing query to the adjunct route server for that portion of the call.
Chaperones	A chaperone is a designated phone user who can announce company policies to the call, monitor the call, and record the call, if required. Chaperone restrictions exist so that the parties that are involved in the call cannot converse without the presence of the chaperone.

Feature	Interaction
Cisco Unified Mobility	Unified Communications Manager allows the route decision from the adjunct route server for the following Cisco Unified Mobility features: • Mobile Voice Access • Enterprise Feature Access • Dial-via-Office Reverse Callback Unified Communications Manager does not send a routing query for the following Cisco Unified Mobility features: • Cell pickup • Desk pickup • Session handoff
Conferences	When a phone user creates a conference, External Call Control may be invoked for the primary call and consultative call.
Directory Numbers	When you configure directory numbers as four- or five-digit extensions (enterprise extensions), you need to configure two translation patterns if on-net dialing supports four or five digits. One translation pattern supports globalizing the calling and called numbers, and a second translation pattern supports localizing the calling and called numbers.
Do Not Disturb	By default, the DND setting for the user takes effect when the user rule on the adjunct route server indicates that the adjunct route server sent a continue obligation. For example, if the adjunct route server sends a continue obligation, and the user has DND-R enabled, Unified Communications Manager rejects the call.
Emergency Call Handling	**Caution** We strongly recommend that you configure a very explicit set of patterns for emergency calls (for example, 911 or 9.11) so that the calls route to their proper destination (for example, to Cisco Emergency Responder or a gateway) without having to contact the route server for instructions on how to handle the call.
Transfer	When a phone user transfers a call, External Call Control may be invoked for both the primary call and consultative call. However, Unified Communications Manager cannot enforce any routing rules from the adjunct route server between the party that transfers and the target of the transfer.

External Call Control Restrictions

Table 74: External Call Control Restrictions

Restriction	Description
Adding Parties	The chaperone cannot use the phone to add parties to a conference after the conference begins, because the call must be put on hold for the chaperone to add parties. The other parties on the conference may add additional parties to the conference. The configuration for the **Advanced Ad Hoc Conference Enabled** service parameter, which supports the Cisco CallManager service, determines whether other parties can add participants to the conference. If the service parameter is set to **True**, other parties can add participants to the conference.
Call Transfer	The chaperone cannot use the phone to transfer the conference call to another party.
Conference Log Out	When the chaperone leaves the conference, the entire conference ends.
Conference Softkey	After the chaperone creates a conference, the **Conference** softkey, if available, is disabled on the phone.
Hold	The chaperone cannot use the phone to put the conference call on hold.
Recording	If the chaperone starts recording before the feature makes a consultative call to the party that will join the conference, Unified Communications Manager suspends recording while the chaperone makes the consultative call; recording resumes after the conference is established.



CHAPTER 60

Configure Call Queuing

- Call Queuing Overview, on page 741
- Call Queuing Prerequisites, on page 743
- Call Queuing Task Flow, on page 743
- Call Queuing Interactions, on page 750
- Call Queuing Restrictions, on page 750
- Performance and Scalability for Hunt Pilots with Call Queuing, on page 751

Call Queuing Overview

Unified Communications Manager provides Call Queuing to place callers in a queue until hunt members are available to answer them. An administrator can set the default so callers receive an initial greeting announcement before the call is extended to an agent or the default can be changed so the initial announcement plays only after the caller is put in the queue followed by Music On Hold or Tone On Hold. If the caller remains in queue for a specified period of time, a secondary announcement is played at a configured interval until the call can be answered or until the maximum wait timer expires.

When an incoming call reaches the hunt pilot, the following functions are provided:

- A caller may be connected to an initial customizable greeting announcement before proceeding.
- If one or more line members are logged in to the hunt pilot and are in an idle state, and if no calls are queued, the call is extended to the line member that has been idle for the longest period of time.
- If no line members answer a call, that caller is not placed in queue. The call is routed to a new destination or disconnected, based on the setting When no hunt members answer, are logged in, or registered.
- If a line member does not answer a queue-enabled call, that line member is logged off the hunt group only if the setting **Automatically Logout Hunt Member on No Answer** is selected in the Line Group setting window.
- Calls are placed in queue only if all members are busy.
- A caller who is waiting in queue may hear Music On Hold and a repeating (customizable) periodic announcement.
- After a line member becomes idle, the caller with the longest wait time across multiple hunt groups is extended to the idle line member. If the idle line member does not answer the call, the caller is returned to the previous position in the queue.

- If a queued call exceeds its maximum wait time or the maximum number of callers allowed in queue is exceeded, the call can be routed to an alternate number or it can be disconnected, depending on how the hunt pilot is configured. The alternate number can be one of the following:
  - A hunt pilot DN with queuing either enabled or disabled
  - A voicemail DN
  - A line DN
  - A shared DN
- Line members can display the queue status of their queue-enabled hunt pilots. The queue status display provides the following types of information:
  - Hunt pilot pattern
  - Number of queued callers on each hunt pilot
  - Longest waiting time

Call queuing works in conjunction with existing hunt pilots, but there are no changes in the behavior of the hunting operation for either queuing or nonqueuing hunt pilots. Hunt pilots that have call queuing enabled provide the following features:

- Queuing-enabled hunt pilot calls can only be received by line members one call at a time. Two queuing-enabled hunt pilot calls cannot be offered to a line member. A line member can receive calls directly to the DN or from non-queuing hunt pilots.
- Line members who do not answer calls that are routed by hunt pilots are automatically logged out. A line member is automatically logged out of a device if the line member receives a queuing-enabled hunt pilot call and does not answer the call before timeout occurs. In the case of a shared-line deployment, all devices configured with the same shared line are logged out. You can configure this behavior from the Line Group setting window by selecting Automatically Logout Hunt Member on No Answer. Line members are logged out only if this check box is checked.

For information about Call Queuing monitoring or announcements monitoring, see *Cisco Unified Real Time Monitoring Tool Administration Guide*.

You can configure the inbound calls to change to the connected call state before playing the queuing announcement while the call is extended to a hunt member in the queuing-enabled hunt pilot.

Secure Call Queuing



Important This section is applicable from Release 14SU2 onwards.

When a secure call is placed to Hunt Pilot and all the Line Groups are busy, callers waiting in the queue hear Music On Hold and repeating (customizable) periodic announcements until a live agent answers the call. During this process, the call is temporarily placed on hold. If the endpoint does not support SRTP fallback, the call placed to Parking Lot (non-secure device) drops off due to the crypto mismatch.

Unified Communications Manager has now enhanced the secure call support to Native Call Queuing, enabling crypto capabilities of a temporary hold call and avoiding call drop-offs. Unified CM handles the originating

Secure Real-Time Transport Protocol (SRTP) only call as a secure call throughout, irrespective of the SRTP fallback option status.

Call Queuing Prerequisites

- Cisco IP Voice Media Streaming (IPVMS) Application, which should be activated on at least one node in the cluster
- Cisco CallManager service that is running on at least one server in the cluster
- Cisco RIS Data Collector service that is running on the same server as the Cisco CallManager service
- Cisco Unified Communications Manager Locale Installer, if you want to use non-English phone locales or country-specific tones

Call Queuing Task Flow

Procedure

	Command or Action	Purpose
Step 1	Configure Announcements, on page 743	Configure announcements through uploading .wav files.
Step 2	Configure Music On Hold, on page 744	Configure Music On Hold (MoH) audio source.
Step 3	Configure Hunt Pilot Queuing, on page 748	Enables call queuing hold option for the calls in a queue until they are answered.
Step 4	Automatically Logout Hunt Member on No Answer, on page 749	Allows line members to log off the hunt list automatically.

Configure Announcements

Cisco Unified Communications Manager allows you to:

- use the existing Cisco-provided announcements,
- change the message or tone that you want an announcement to play,
- insert custom announcement .wav files,
- assign the locale for the announcement,
- change the description for the announcement,
- change the message or tone that you want an announcement to play.

Feature announcements are used by specific features such as Music On Hold (MoH) in association with Hunt Pilot call queuing or External Call Control.

There are up to 50 feature announcements available. These announcements can be Cisco-provided audio files or uploaded custom .wav files.

All custom announcement .wav files must be uploaded to all servers in the cluster.

Procedure

Step 1 In Cisco Unified Communications Manager, select **Media Resources** > **Announcements**.
The **Find and List Announcements** window displays.

Step 2 Select a hyperlink to the announcement you want to use.

Example:

Hyperlink—Wait_In_Queue_Sample
You can edit the announcement description or choose a customized announcement if uploaded.

Step 3 To upload a .wav file to use as a custom announcement, click **Upload File**.
The **Upload File** window opens.

Step 4 In the **Upload File** window, choose the locale, enter the filename or browse to select the .wav file, and click **Upload File**.

The upload process begins, and may take a few minutes depending on the file. The Status is updated after processing is complete.

Step 5 Click **Close** to close the upload window.
The **Announcement Configuration** window refreshes to update the uploaded file status.

Step 6 To play the customized announcement, ensure that the **Enable** check box is checked in the Announcement by Locale pane in the **Announcements Configuration** window.

Step 7 After you make the changes in the **Announcements Configuration** window, click **Save**.

What to do next

You must upload the announcement on each node in the cluster, because the announcement files are not propagated between servers in a cluster. Browse to Cisco Unified Communications Manager Administration on each server in the cluster and repeat the upload process.

Configure Music On Hold

You can configure Music On Hold (MoH) to play an optional initial greeting announcement when a caller is first put on hold and also to play a periodic repeating announcement. These announcements can use one of the Cisco-provided audio files or a file that is uploaded into the system.

Perform the following procedure to add or update a Music On Hold audio source, to associate an existing audio source with an audio stream number, or to upload a new custom audio source.

Procedure

Step 1 From the Cisco Unified Communications Manager, choose **Media Resources** > **Music On Hold Audio Source**.

The **Find and List Music On Hold Audio Sources** window appears.

Step 2 To add a new Music On Hold audio source, click **Add New**. To update a Music On Hold audio source, locate a specific Music On Hold audio source. Based on the search criteria you specify, the system displays search results for the record that matches all the criteria.

Step 3 Enter the appropriate settings, as described in Audio Source Fields for Music On Hold, on page 745.

Step 4 Click **Save.**

The list box at the bottom of the window shows the new Music On Hold audio source. The MOH Audio Source File Status pane shows the MOH audio translation status for the added source.

Audio Source Fields for Music On Hold

Table 75: Music On Hold Audio Source Information

Field	Description
MOH Audio Stream Number	Use this field to choose the stream number for this MOH audio source. Click the drop-down list and choose a value from the list. For existing MOH audio sources, the value appears in the MOH Audio Source title.
MOH Audio Source File	Use this field to choose the file for this MOH audio source. Click the drop-down list and choose a value.
MOH Audio Source Name	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.
Allow Multicasting	Check this check box to specify that the selected MOH audio source allows multicasting.

Field	Description
MOH Audio Source File Status	This pane displays the following information about the source file for the selected MOH audio source: • InputFileName • ErrorCode • ErrorText • DurationSeconds • DiskSpaceKB • LowDateTime • HighDateTime • OutputFileList • MOH Audio Translation completion date **Note** OutputFileList includes information on ULAW, ALAW, G.729, and Wideband wav files and status options.

Table 76: Announcement Settings

Field	Description
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 **Allow Multi-casting** is not checked and the **Initial Announcement for queuing-enabled hunt pilot calls** field is set to **Play announcement if call is queued**. • Played by ANN if **Allow Multi-casting** check box is checked or if **Initial Announcement for queuing-enabled hunt pilot calls** is set to **Play announcement before routing to Hunt Member**.

Field	Description
Initial Announcement for queuing-enabled hunt pilot calls	Choose one of the following to determine when to play the initial announcement: • Play announcement before routing to Hunt Member • Play announcement if call is queued
Periodic Announcement	Choose a periodic announcement from the drop-down list. **Note** To select MoH with no periodic announcement, choose the **Not Selected** option. Click the **View Details** link to view the following Periodic Announcement information: • Announcement Identifier • Description • Default Announcement **Note** The MOH server always plays the periodic announcement regardless of other settings.
Periodic Announcement Interval	Enter a value (in seconds) that specifies the periodic announcement interval. Valid values are 10 to 300. The default value is 30.
Locale Announcement	Locale Announcement depends upon the locale installation package that has been installed. **Note** • Prompts played by MOH will use the setting for Locale Announcement. • Prompts played by ANN will use the User Locale of the calling party.

Table 77: Music On Hold Audio Sources

Field	Description
(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.

Field	Description
Upload File	To upload an MOH audio source file that does not appear in the drop-down list, click **Upload File**. In the **Upload File** window, either enter the path of an audio source file or navigate to the file by clicking **Browse**. After you locate the audio source file, click the **Upload File** button to complete the upload. After the audio file gets uploaded, the Upload Result window displays the result of the upload. Click **Close** to close this window. **Note** When you upload a file, the file is uploaded to the Unified Communications Manager server and performs audio conversions to create codec-specific audio files for MOH. Depending on the size of the original file, processing may take several minutes to complete. **Note** Uploading an audio source file to an MOH server uploads the file only to one MOH server. You must upload an audio source file to each MOH server in a cluster by using Cisco Unified Communications Manager Administration on each server. MOH audio source files do not automatically propagate to other MOH servers in a cluster.

Configure Hunt Pilot Queuing

When a hunt pilot has more calls distributed through the call distribution feature than its hunt members can handle at any given time, call queuing holds these calls in a queue until they can be answered.

When queuing is enabled, both Forward Hunt No Answer and Forward Hunt Busy are automatically disabled. Conversely, if Forward Hunt No Answer or Forward Hunt Busy is enabled, queuing is automatically disabled.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **Call Routing** > **Route/Hunt** > **Hunt Pilot** to configure hunt pilots.

Step 2 Select the hunt pilot that you need to configure for Queuing.

Step 3 Navigate to the Queuing section of the **Hunt Pilot Configuration** window.

Step 4 Check the **Queue Calls** check box to enable queuing.

Step 5 Choose a Music On Hold (MoH) source from the drop-down list box to be used to play announcements and provide queue hold treatments.

The MoH source can be configured as unicast or multicast. The caller-side Media Resource Group List (MRGL) takes precedence for multicast or unicast.

If you do not select a source, the default Network Hold MoH/MoH Source and Announcements is used.

The MoH source announcement locale is used to determine the language used for the announcement. Only one type of language announcement can be played per hunt pilot.

Step 6 In the **Maximum Number of Callers Allowed in Queue** field, enter an integer value for the number of callers allowed in the queue for this hunt pilot.

The default value is 32. The field range is from 1 to 100.

Step 7 Choose one of the following options when the maximum number of callers in the queue is reached:

- If you want subsequent calls to be disconnected, select **Disconnect the call**.
- If you want subsequent calls to be routed to a secondary destination, select **Route the call to this destination**. Provide a specific device DN, shared line DN, or another hunt pilot DN.
- (Optional) You may also select **Full Queue Calling Search Space** from the drop-down list. Used to determine which partition to search when attempting to complete a call.

Step 8 In the Maximum Wait Time in Queue field, enter an integer value to set the maximum wait time, in seconds, in a queue.

The default value is 900 seconds. The field range is from 10 to 3600 seconds.

Step 9 Choose one of the following options when the maximum wait time is reached:

- If you want that call to be disconnected, select **Disconnect the call**.
- If you want that call to be routed to a secondary destination, select **Route the call to this destination**. Provide a specific device DN, shared line DN, or another hunt pilot DN.
- (Optional) You may also select **Maximum Wait Time Calling Search Space** from the drop-down list. Used to determine which partition to search when attempting to complete a call.

Step 10 When no line members are logged in or registered at the time of an incoming call, choose one of the following options:

- If you need that call to be disconnected, select **Disconnect the call**.
- If you need that call to be routed to a secondary destination, select **Route the call to this destination**. Provide a specific device DN, shared line DN, or another hunt pilot DN.
- (Optional) You may also select **No hunt members logged in or registered Calling Search Space** from the drop-down list. Used to determine which partition to search when attempting to complete a call.

Step 11 Click **Save**.

Automatically Logout Hunt Member on No Answer

Allows line members to log off the hunt list automatically. If an agent does not answer a queuing-enabled hunt pilot call, that agent will be logged off of the hunt group and will not receive additional hunt pilot calls unless he presses the "HLOG" soft key on the phone to log into the hunt pilot.

Line members can log back in using the "HLOG" softkey or PLK.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **Call Routing** > **Route/Hunt > Line Group** to configure line groups.

Step 2 Choose the line group that you need to configure from the **Find and List Line Groups** window.

Step 3 Navigate to the Hunt Options section of the **Line Group Configuration** window.

Step 4 Ensure that the **Automatically Logout Hunt Member on No Answer** check box is checked.

Step 5 Click **Save**.

Call Queuing Interactions

Feature	Interaction
SIP Rel1XX Options	If a call is routed to a queuing-enabled hunt pilot through SIP ICT, the SIP ICT uses the SIP profile that has SIP Rel1XX Options set to **Send PRACK if 1XX contains SDP**. As a result, the initial announcement is played to every call before the call is extended to the line member. The above existing interaction for SIP ICT does not apply if **Connect Inbound Call before Playing Queuing Announcement** checkbox is checked under **DeviceDevice Settings SIP Profile** > **Trunk Specific Configuration** in Cisco Unified CM Administration. If **Connect Inbound Call before Playing Queuing Announcement** checkbox is not checked the interaction for SIP ICT remains the same. However, it does not guarantee the initial announcement can always be heard by a caller from the PSTN side. The initial announcement will not be heard by a caller from the PSTN side if the PSTN provider doesn't open the voice path until a Connect message is received on the call.
Hunt Pilots and Hunt Groups	• The logoff notification functionality for hunt groups changes when Call Queuing is enabled for a hunt pilot. If Call Queuing is enabled for a hunt pilot, the Hunt Group Logoff Notification does not play when users log out of a hunt group or are logged off because they missed their turn in the queue. • If the hunt list has multiple line groups, these line groups must have the same setting for **Automatically Logout Hunt Member on No Answer**. • Hunt Pilot still queues calls, even when all hunt members are logged out.The line group members should not be added in more than one line group and even if they are added in second line groups, those second line groups should not be in the same Hunt list. • All hunt options must be set to Try Next Member, then Try Next Group in the hunt list.

Call Queuing Restrictions

The following general restrictions apply to call queuing:

- H.323 Fast Start does not support Call Queuing.
- Queue status PLK is supported only with the following LCD display phones for both SCCP and SIP: 6921, 6941, 6945, 6961, 7911G, 7931G, 7942G, 7945G, 7962G, 7965G, 7975G, 8961, 8945, 8941, 9951, 9971, 7800 and 8800 series.

- Log Out of Hunt Groups (HLog) is not compatible with Cisco Extension Mobility Cross Cluster (EMCC); Call Queuing should not be deployed with EMCC.
- Unified Communications Manager does not support Unified Mobility with Call Queuing.
- In a H323 to SIP interworking scenario, the user may not hear initial announcement, MoH, periodic announcement or observe call failure in a native call queuing flow due to interworking delays. In such a scenario it is advised to use only SIP protocol.

Performance and Scalability for Hunt Pilots with Call Queuing

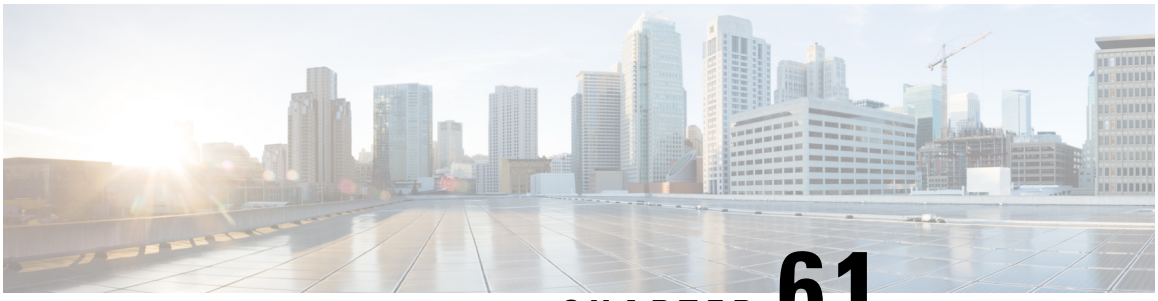
The following performance and scalability restrictions apply:

- A single Unified CM Cluster supports a maximum of 15,000 hunt list devices.
- A single Unified CM Subscriber supports a maximum of 100 hunt pilots with call queuing enabled per node
- Hunt list devices may be a combination of 1500 hunt lists with ten IP phones in each hunt list, 750 hunt lists with twenty IP phones in each hunt list, or similar combinations



Note When using the broadcast algorithm for call coverage, the number of hunt list devices is limited by the number of busy hour call attempts (BHCA). Note that a BHCA of 10 on a hunt pilot pointing to a hunt list or hunt group containing 10 phones and using the broadcast algorithm is equivalent to 10 phones with a BHCA of 10.

- The maximum number of hunt pilots is 100 per Unified CM subscriber node with call queue enabled when configured with 32 callers which is allowed in the queue. The total number of queue slots per node (the value of "Maximum Number of Callers Allowed in Queue" for all Call Queuing Enabled Hunt Pilots on the node combined) is limited to 3200. The maximum number of simultaneous callers in a queue for each hunt pilot is 100, meaning 100 callers per hunt pilot is allowed in a queue and the maximum number of hunt pilots is reduced to 32. The maximum number of members across all hunt lists does not change when call queuing is enabled.
- The maximum wait time in queue for each hunt pilot that you can configure ranges from 0 to 3600 seconds (default 900). An increase in the number of hunt lists can require you to increase the dial plan initialization timer that is specified in the Unified Communications Manager service parameters. We recommend that you set the dial plan initialization timer to 600 seconds if you have 1500 hunt lists configured.
- We recommend having no more than 35 directory numbers for a single line group when using broadcast algorithms with call queuing. Additionally, the number of broadcast line groups depends on the busy hour call completion rate (BHCC). If there are multiple broadcast line groups in a Unified CM system, the number of maximum directory numbers in a line group must be less than 35. The number of busy hour call attempts (BHCA) for all the broadcast line groups should not exceed 35 calls set up per second.



CHAPTER 61

Configure Call Throttling

- Call Throttling Overview, on page 753
- Call Throttling Configuration Task Flow, on page 754

Call Throttling Overview

Call Throttling allows your system to automatically throttle or deny new call attempts. The system takes this action when conditions cause users to experience a delay in the interval between going off hook and receiving a dial tone.

Some factors that can result in this delay are as follows:

- Heavy call activity
- Low CPU availability
- Routing loops
- Disk I/O limitations
- Disk fragmentation

The system uses the values that are specified in the call throttling parameters to determine a possible delay to dialtone and also to determine when conditions no longer require call throttling.

When throttling is necessary to prevent excessive delay to dialtone, the system enters a Code Yellow state and new call attempts are throttled (denied).

When the system calculates the delay to dialtone as being over the threshold that is configured in the call throttling service parameters, Unified Communications Manager rejects new calls. When call throttling activates, a user who attempts a new call receives a reorder tone and, depending on the phone model, may also receive a prompt on the phone display.

Call throttling effectively prevents the type of excessive delays that can cause a user to complain to the system administrator or question whether the system is down or the phone is broken. Your system constantly monitor the system to anticipate when such latency could occur.

When the delay to dialtone is within the guidelines of the call throttling service parameters, Unified Communications Manager stops throttling calls by exiting the Code Yellow state and new calls are again allowed.

Call Throttling Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Configure Call Throttling, on page 754	Enables Call throttling automatically when your system detects conditions such as heavy call activity, low CPU availability, and disk fragmentation.
Step 2	Configure Memory Throttling, on page 755	Configures memory throttling for your system.

Configure Call Throttling

Call throttling occurs automatically when your system detects conditions such as heavy call activity, low CPU availability, and disk fragmentation. The system automatically exits throttling when these conditions are fixed. Call Throttling is configured via advanced service parameters. For many deployments, the default settings are sufficient.

Caution We recommend that you not modify call throttling parameters unless advised to do so by customer support.

Procedure

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

Step 2 From the **Server** drop-down list, choose a server.

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

Step 4 Click **Advanced**.

Step 5 Under **Call Throttling**, configure values for the cll throttling service parameters. For parameter help descriptions, click the parameter name in the GUI.

- Code Yellow Entry Latency
- Code Yellow Exit Latency Calendar
- Code Yellow Duration
- Max Events Allowed
- System Throttle Sample Size

Step 6 Click **Save**.

Configure Memory Throttling

Use this procedure to configure memory throttling for your system.

Procedure

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

Step 2 From the **Server** drop-down list, select a Unified Communications Manager server.

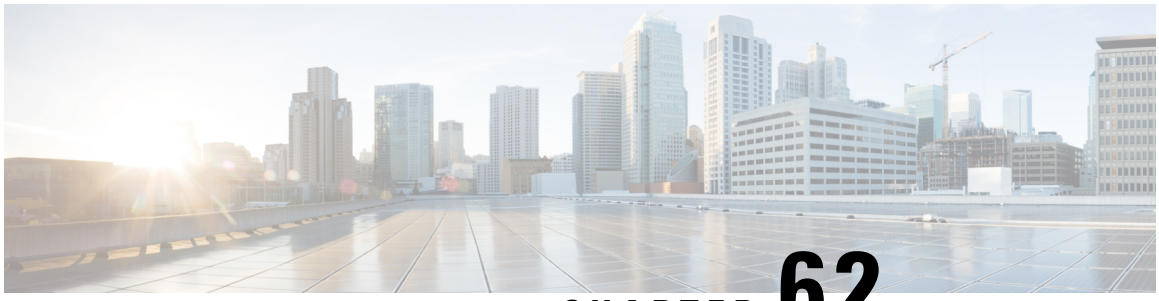
Step 3 From the **Service** drop-down list, select **Cisco CallManager**.

Step 4 Click **Advanced**.

Step 5 Set the **Enable Memory Throttling** parameter to **True**.

Step 6 Configure values for the additional service parameters in the **Memory Throttling** area. For parameter help, click the parameter name in the GUI.

Step 7 Click **Save**.



CHAPTER 62

Configure Logical Partitioning

- Logical Partitioning Overview, on page 757
- Logical Partitioning Configuration Task Flow, on page 757
- Logical Partitioning Interactions, on page 764
- Logical Partitioning Restrictions, on page 765

Logical Partitioning Overview

With logical partitioning, you can support PSTN and VoIP calls on a single system while meeting regulatory requirements for call separation. For example, under regulatory constraints in India, all calls that are received from or sent to an external phone must be handed off to and carried by a local or long-distance service provider over the full length of the connection, with the applicable toll charges. You can create a single Unified Communications Manager cluster that routes calls appropriately to the PSTN or the VoIP network according to the caller's location and the phone number being called.

logical partitioning defines which sets of VoIP devices are allowed to communicate with each other. Users do not have to remember to use one line for PSTN and one line for VoIP. Phones making off-net calls are only allowed to talk to a PSTN gateway. It's like having two networks to separately handle your VoIP and PSTN calls, but without the expense of dual infrastructure.

Logical Partitioning Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Enable Logical Partitioning, on page 758	Enable Logical Partitioning.
Step 2	To Configure Geolocations, on page 758, perform the following subtasks: • Create Geolocations, on page 759 • Assign Geolocations, on page 759 • Set the Default Geolocation, on page 760	Configuring geolocations is a two-step process: defining locations and assigning them to devices. You also can set the default location to be used by all devices in the cluster.

	Command or Action	Purpose
Step 3	Configure a Logical Partitioning Default Policy, on page 760	Set up a default policy for devices that are not associated with a geolocation or geolocation filter. The policy allows or denies PSTN calls between these devices.
Step 4	Configure Devices to Avoid Logical Partitioning Checks, on page 761	You can specifically exempt devices and device pools from the partitioning checks.
Step 5	To Configure Geolocation Filters, on page 761, perform the following subtasks: • Create Geolocation Filter Rules, on page 762 • Assign Geolocation Filters, on page 762 • Set the Default Geolocation Filter, on page 762	Logical partitioning assigns a unique identifier to each device based on its location. When one device calls another, these identifiers are used to determine whether the call is allowed and what routing is appropriate. You can choose which fields are used to create this identifier. For example, you can apply different policies based on the room or floor within a building.
Step 6	Define a Set of Logical Partitioning Policy Records, on page 763	Define a set of logical partitioning policies for allowing or denying calls between geolocations. Before calls between geolocations are allowed to proceed, the system checks to be sure that calls are allowed between the specified geolocations based on these policies.
Step 7	(Optional) Enable Location Conveyance, on page 763	Configure location conveyance if you want to communicate geolocation information about devices across clusters.

Enable Logical Partitioning

Use this procedure to turn on the Logical Partitioning feature.

Procedure

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

Step 2 For the **Enable Logical Parititioning** enterprise parameter, choose **True** from the drop-down list.

Step 3 Click **Save**.

Configure Geolocations

Configuring geolocations is a two-step process: defining locations and assigning them to devices. You also can set the default location to be used by all devices in the cluster.

Procedure

	Command or Action	Purpose
Step 1	Create Geolocations, on page 759	Configure geolocations to specify geographic locations. These are used to associate devices with regulatory features such as logical partitioning. Geolocations are used in policy decisions, such as in-country regulations.
Step 2	Assign Geolocations, on page 759	Assign a geolocation to a device or device pool.
Step 3	Set the Default Geolocation, on page 760	Specify a default geolocation for all devices and device pools in this cluster.

Create Geolocations

Use this procedure to create geolocations that you can assign to the devices in your system. You can use the geolocations for logical partitioning.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Geolocation Configuration**.

Step 2 Click **Add New**.

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

Step 4 Configure the fields on the **Geolocation Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 5 Click **Save**.

Step 6 Repeat this procedure to create additional geolocations.

Assign Geolocations

Assign a geolocation to a device or device pool.

Procedure

Step 1 From Cisco Unified CM Administration, choose one of the following menu items:

- **Device** > **Phone**
- **Device** > **Trunk**
- **Device** > **Gateway**
- **System** > **Device Pool**

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing device or device pool. Enter search criteria, and then choose an existing device or device pool from the resulting list.
- Click **Add New** to add a new device or device pool. For devices, choose device types and protocols as needed and click **Next**.

Step 3 From the Geolocation drop-down list, choose a geolocation that you configured.

Step 4 Click **Save**.

Set the Default Geolocation

Specify a default geolocation for all devices and device pools in this cluster.

Procedure

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

Step 2 From the **Default Geolocation** drop-down list, choose a Geolocation that you configured. The default value is **Unspecified**.

Step 3 Click **Save**.

Step 4 Click **Apply Config**.

Step 5 (Optional) If you need to override this default for a specific device or device pool, enter the value on either the **Device Configuration** or **Device Pool Configuration** window, and then click **Save**.

Configure a Logical Partitioning Default Policy

Set up a default policy for devices that are not associated with a geolocation or geolocation filter. The policy allows or denies PSTN calls between these devices.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Logical Partitioning Policy Configuration**

Step 2 Click **Add New**.

Step 3 Configure the fields on the **Logical Partition Policy Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 4 Click **Save**.

Note
If a policy that contained the value Allow is then later changed to Deny, then it remains Deny. The opposite is also true. A policy previously set to Deny, later changed to Allow is an Allow. The **Cisco Unified Reporting** > **Geolocation Policy Report** can help you identify policies that overlap.

Configure Devices to Avoid Logical Partitioning Checks

You can specifically exempt devices and device pools from the partitioning checks.

Procedure

Step 1 From Cisco Unified CM Administration, choose one of the following menu items:

- **Device** > **Phone**
- **Device** > **Trunk**
- **Device** > **Gateway**
- **System** > **Device Pool**

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing device or device pool. Enter search criteria and then choose an existing device or device pool from the resulting list.
- Click **Add New** to add a new device or device pool. For devices, choose device types and protocols as needed and click **Next**.

Step 3 From the **Geolocation** drop-down list, choose **Unspecified**.

Step 4 Click **Save**.

Configure Geolocation Filters

Logical partitioning assigns a unique identifier to each device based on its location. When one device calls another, these identifiers are used to determine whether the call is allowed and what routing is appropriate. You can choose which fields are used to create this identifier. For example, you can apply different policies based on the room or floor within a building.

Procedure

	Command or Action	Purpose
Step 1	Create Geolocation Filter Rules, on page 762	Geolocation filters allow you to specify which fields are used to create a geolocation identifier. This feature is used to make policy decisions on a subset of the geolocation objects.
Step 2	Assign Geolocation Filters, on page 762	
Step 3	Set the Default Geolocation Filter, on page 762	Configure the Default Geolocation Filter enterprise parameter to specify a default geolocation filter for a cluster. This parameter determines the default geolocation filter setting for all devices and device pools that are not associated with a geolocation filter.

Create Geolocation Filter Rules

Use this procedure to create geolocation filters that you can use for logical partitioning decisions.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Geolocation Filter**.

Step 2 Click **Add New**.

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

Step 4 Check the check boxes that correspond to the items you want to use for logical partitioning decisions.

Step 5 Configure the fields on the **Geolocation Filter Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 6 Click **Save**.

Step 7 Repeat these steps to create additional geolocation filters.

Assign Geolocation Filters

Procedure

Step 1 From Cisco Unified CM Administration, choose one of the following menu items:

- **Device** > **Phone**
- **Device** > **Trunk**
- **Device** > **Gateway**
- **System** > **Device Pool**

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing device or device pool. Enter search criteria and then choose an existing device or device pool from the resulting list.
- Click **Add New** to add a new device or device pool. For devices, choose device types and protocols as needed and click **Next**.

Step 3 From the **Geolocation Filter** drop-down list, choose a geolocation filter that you configured.

Step 4 Click **Save**.

Set the Default Geolocation Filter

Procedure

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

Step 2 From the **Default Geolocation** drop-down list, choose a Geolocation that you configured. The default value is **Unspecified**.

Step 3 Click **Save**.

Step 4 Click **Apply Config**.

Step 5 (Optional) If you need to override this default for a specific device or device pool, specify the default geolocation filter value on either the **Device Configuration** or **Device Pool Configuration** window, and then click **Save**.

Define a Set of Logical Partitioning Policy Records

Define a set of logical partitioning policies for allowing or denying calls between geolocations. Before calls between geolocations are allowed to proceed, the system checks to be sure that calls are allowed between the specified geolocations based on these policies.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Logical Partitioning Policy Configuration**.

Step 2 Perform one of the following tasks:

- Click **Find** to modify the settings for an existing logical partitioning policy. Enter search criteria and then choose an existing logical partitioning policy from the resulting list.
- Click **Add New** to add a new logical partitioning policy.

Step 3 Configure the fields on the **Logical Partitioning Policy Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Note
If any policy is left blank without any configuration values, it will become a blank geolocation policy and configuring a Logical Policy for a specific Device Type with the blank Logical Partitioning configurations makes Unified Communications Manager add the policy value (Allow or Deny) in the configured device type.

Step 4 Click **Save**.

Enable Location Conveyance

Location Conveyance is an optional configuration that lets you share geolocation information across clusters.

Procedure

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

Step 2 Do one of the following:

- Click **Find** and select an existing trunk.
- Click **Add New** to configure a new trunk.

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

Step 4 In the **Geolocation Information** area, select a **Geolocation** and **Geolocation Filter**.

Step 5 To enable Location Conveyance, check the **Send Geolocation Information** check box.

Step 6 Click **Save**.

Logical Partitioning Interactions

Table 78: Logical Partitioning Interactions

Feature	Interaction
Ad Hoc Conference, Join, Join Across Lines, Call Forwarding, Call Transfer	Logical partitioning handling does not take place in the following circumstances: • When all participants are VoIP phones. • When the geolocation or geolocation filter does not associate with a device.
Barge, cBarge, and Remote Resume	Logical partitioning handling does not take place in the following circumstances: • When both the caller and the callee devices are VoIP phones, logical partitioning policy checks are ignored. • For the participants in cBarge/Barge, no logical partitioning policy checking exists, and you cannot prevent logical-partitioning-denied scenarios.
Cisco Unified Mobility	Logical partitioning handling does not take place in the following circumstances: • Geolocation or geolocation filter does not associate with the involved devices. • No logical partitioning support exists when a dual-mode phone is used.
CTI Handling	Logical partitioning handling does not take place in the following circumstances: • When a geolocation or geolocation filter does not associate with any device, handling does not occur. • When all the involved devices specify VoIP phones, handling does not occur.
Extension Mobility	Logical partitioning handling does not take place in the following circumstances: • A geolocation or geolocation filter does not associate with a VoIP phone that is logged on to Cisco Extension Mobility, nor does it associate with the calling party or called party device. • The VoIP phone that is logged on to Cisco Extension Mobility calls or receives a call from a VoIP phone.

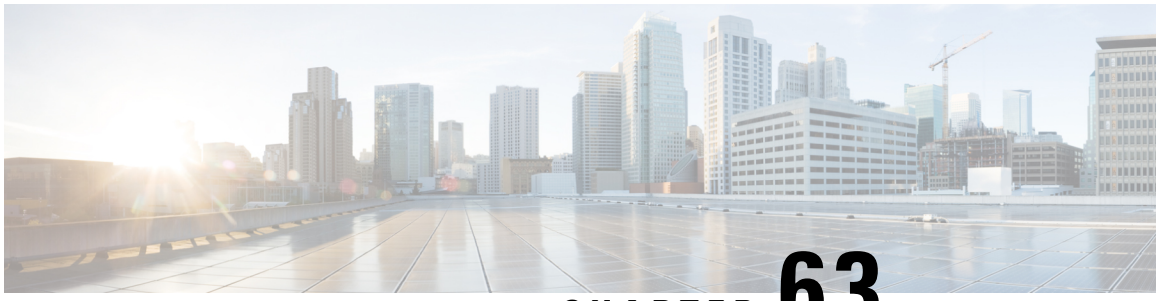
Feature	Interaction
Meet-Me Conference	Logical partitioning handling does not take place in the following circumstances: • When all participants are VoIP phones, handling does not occur. • When geolocation or geolocation filter does not associate with a device, no policy check takes place for that device.
Route Lists and Hunt Pilots	Logical partitioning handling does not take place in the following circumstances: • When both the calling party and called party devices are VoIP phones, handling does not occur. • All devices must associate with both a geolocation and geolocation filter. If any device does not associate with both geolocation and geolocation filter, handling does not occur.
Shared Line	Logical partitioning handling does not take place in the following circumstances: • When both the caller and the callee devices are VoIP phones, no handling occurs. • When geolocation or geolocation filter does not associate with any device, no handling occurs.

Logical Partitioning Restrictions

Table 79: Logical Partitioning Restrictions

Restriction	Description
Barge/cBarge	Barge/cBarge does not occur; the call instance is dropped. For the participants in cBarge/Barge, no logical partitioning policy checking exists, and you cannot prevent logical-partitioning-denied scenarios.
BLF Presence	BLF Presence notifications are not checked for a logical partitioning policy.
Cisco Extension Mobility	When Cisco Extension Mobility logs in to a phone in a different geolocation, outgoing PSTN calls can occur when Local Route Groups are configured. Incoming PSTN calls are not placed to the phone but receive a reorder tone.
Cisco Unified MeetingPlace	The system does not support the logical partitioning feature for calls that involve Cisco Unified MeetingPlace or Cisco Unified MeetingPlace Express.
Conferences	The logical partitioning checks are not supported for participants across conferences in conference chaining. For example, meet-me and adhoc chained conferences can have participants that are logical partitioning denied.

Restriction	Description
H.225 gatekeeper-controlled trunk	Cisco Unified Communications Manager does not communicate geolocation information over a H.225 gatekeeper-controlled trunk.
H.323 and MGCP Gateways	Cisco Unified Communications Manager does not communicate geolocation info to H.323 or MGCP gateways. Communication to a SIP gateway can be disabled through the SIP trunk check box.
Mobility Cell Pickup	Logical partitioning deny handling takes place after call is answered on the mobile phone. The logical partitioning policy check does not occur before the call is placed to the mobile phone (as it happens for a basic SNR call). The system checks the logical partitioning policy after the mobile phone answers the call.
Q.SIG intercluster trunk	Intercluster trunks (ICT) with the Q.SIG protocol are not allowed to communicate geolocation inforomation for the caller or receiving device. The ICT configuration for "Send Geolocation Information" is disabled when the Q.SIG tunneled protocol is selected.
Reorder Tones	No reorder tone (fast busy tone) is provided on IOS H.323 and SIP gateways upon release of connected calls due to logical partitioning policies.
Shared Line Active Call	For a restricted logical partitioning scenario, the shared line drops the active call information for the duration of the call, even if a feature moves the shared-line call to the allowed category.
User Agent Server	The logical partitioning policy checks in the logical partitioning-aware cluster that receives this geolocation may cancel the call if the policy is denied.



CHAPTER 63

Configure Location Awareness

- Location Awareness Overview, on page 767
- Enhanced Location Awareness of Wireless Access Points, on page 769
- Location Awareness Prerequisites, on page 770
- Location Awareness Configuration Task Flow, on page 770

Location Awareness Overview

Important Meraki Access Points support for Location Awareness is applicable only from Release 12.5(1)SU6 onwards and Release 14SU1 onwards.

Location Awareness allows administrators to determine the physical location from which a phone connects to the company network. For wireless networks, you can view the wireless access point infrastructure, and which mobile devices currently associate to those access points. For wired networks, you can view the Ethernet switch infrastructure and see which devices are currently connected to those switches. This allows you to determine the building, floor, and cube from which a call was placed.

Note Currently, wired phones do not support Location Awareness.

You can view your network infrastructure from **Cisco Unified CM Administration** > **Advanced Features** > **Device Location Tracking Services** > **Switches and Access Points** > **Find and List Switches and Access Points** window.

This feature updates the Unified Communications Manager database dynamically with the following information:

- Network infrastructure devices such as switches and wireless access points, including IP addresses, hostnames, and Basic Service Set Identifier (BSSID) information (where applicable) for each infrastructure device.
- Associated endpoints for each infrastructure device, including:
  - For wireless networks, the list of devices that are currently associated to a wireless access point.

- For wired networks, the list of devices and device types that are currently connected to an ethernet switch.

Cisco Emergency Responder Integration

Location Awareness helps integrated applications such as Cisco Emergency Responder to determine the physical location of a user who places an emergency call. When Location Awareness is enabled, Cisco Emergency Responder learns of a new device to infrastructure association within minutes of a mobile device associating with a new wireless access point, or a desk phone being connected to a new ethernet switch.

When Cisco Emergency Responder first starts up, it queries the Unified Communications Manager Database for the current device to network infrastructure associations. Every two minutes following, the Cisco Emergency Responder checks for updates to the existing associations. As a result, even if a mobile caller places an emergency call while in a roaming situation, Cisco Emergency Responder can quickly determine the physical location of the caller and send emergency services to the appropriate building, floor, or cube.

Wireless Network Updates

To enable Location Awareness for your wireless infrastructure, you can configure Unified Communications Manager to synchronize with a Cisco Wireless LAN Controller. You can synchronize Unified Communications Manager with up to fifty controllers. During the synchronization process, Unified Communications Manager updates its database with the access point infrastructure that the controller manages. In Cisco Unified CM Administration, you can view the status for your wireless access points, including the list of mobile clients that are associated to each access point.

As mobile clients roam between access points, SIP and SCCP signaling from the endpoint communicates the new device to access point association to Unified Communications Manager, which updates its database. Cisco Emergency Responder also learns of the new association by querying the Unified Communications Manager database every few minutes for new endpoints that have changed their association. As a result, if a mobile client places an emergency call, Cisco Emergency Responder has accurate information on the physical location of the user whom placed the call.

If you have a regular synchronization schedule for your Wireless Access Point controllers, Unified Communications Manager adds and updates access points from the database dynamically following each synchronization.

Using Bulk Administration to insert Access Points

If you are using a third-party wireless access point controller, or if you want to export your access points from Cisco Prime Infrastructure, you can use the Bulk Administration Tool to bulk insert your wireless access point infrastructure from a CSV file into the Unified Communications Manager database. Following the bulk insert, the next location update from the mobile device updates the database with the current access point association.

However, Bulk Administration does not allow you to update your access point infrastructure dynamically as new access points get added to your wireless network. If a mobile call gets placed through an access point that was added after the bulk insert, that access point will not have a record in the database, Unified Communications Manager will not be able to match the BSSID of the new access point, and will mark the infrastructure for the wireless device as UNIDENTIFIED AP.

For detailed information on the Bulk Administration Tool, refer to the "Manage Infrastructure Devices" chapter of the *Bulk Administration Guide for Cisco Unified Communications Manager*.

Supported Endpoints for Location Awareness

The following endpoints support tracking via Location Awareness:

- Cisco Uniifed Wireless IP Phone 7925G
- Cisco Unified Wireless IP Phone 7925G-EX
- Cisco Unified Wireless IP Phone 7926G
- Cisco Jabber clients—supported as of 12.5(1)SU1
- Cisco Wireless IP Phone 8821—supported as of 12.5(1)SU1
- Webex App—supported as of 12.5(1)SU1

These endpoints provide upstream infrastructure information, such as BSSID, through Station Info messages to Cisco Unified Communications Manager. Cisco Emergency Responder uses AXL Change Notifications to track these devices through the associated access point.

For device tracking to work, wireless access points must be defined in Cisco Unified Communications Manager. You can do this by syncing a wireless access point controller or using Bulk Administration to import wireless access point infrastructure.

Supported Meraki Wireless Access Point for Location Awareness



Important This section is applicable only until Release 15SU4.

Unified Communications Manager supports Meraki Wireless Access Point for Location Awareness.

Support is limited to the model with this specific OUI. While a model may have multiple OUIs, from the Unified Communications Manager or Emergency Responder perspective, only those OUIs explicitly mentioned are supported. Any other OUIs for the same model are not supported within Unified Communications Manager or Emergency Responder. Refer to the 'Supported Meraki Wireless Access Point for Location Awareness' section in the CER Release Notes documentation for the supported Meraki Wireless Access Points.

Enhanced Location Awareness of Wireless Access Points



Important This section is applicable from Release 15SU5 onwards. With the introduction of this feature, the administrator can see the following changes:

- Unified Communication Manager allows admins to see exact SSID mappings for registered endpoints.
- Each SSID is treated as an individual entry in the Unified Communication Manager.
- Admins must group entries by access point during deletion of an access point.

Key Technical Enhancements

- **Expanded Capacity:** The limit for the infrastructure device database has been increased from 50,000 to 100,000 records to accommodate the higher volume of BSSIDs per Access Point.
- **Bulk Administration Tool (BAT) Updates:** Admins can now upload multiple BSSIDs per AP via CSV, streamlining the provisioning process for large-scale deployments.
- **External Utility Support:** A new utility is available to help administrators extract and format data from the Meraki Dashboard into a BAT-compatible CSV file, reducing manual overhead.
- **CER Integration:** CER has been updated to support one-to-one mapping between individual BSSIDs and Emergency Response Locations (ERLs). This ensures accuracy even when multiple Service Set Identifiers (SSIDs) are broadcast from a single Access Point.

From this release onwards, we additionally support the following hardware:

- Cisco Catalyst 9100 Series Access Points
- Cisco Meraki MR Series Access Points
- Cisco Meraki CW Series Access Points

Deployment Considerations

- **Brownfield Migrations:** Existing configurations will continue to function using legacy logic, while new configurations will utilize the updated, non-normalized storage method.
- **Administrator Experience:** Because each BSSID is now stored as a unique entry, administrators will see multiple records per Access Point in the UI. This provides more granular control over ERL mapping at the SSID level.
- **Utility Usage:** For Meraki environments, we recommend using the provided external utility to derive all BSSIDs from meraki dashboard.

Location Awareness Prerequisites

This feature allows you to synchronize the Cisco Unified Communications Manager database with multiple Cisco Wireless LAN Controllers. You must also set up your Cisco Wireless LAN Controller hardware and your infrastructure of access points. For details, see your controller documentation.

Location Awareness Configuration Task Flow

Complete the following tasks to set up Location Awareness in Cisco Unified Communications Manager.

Procedure

	Command or Action	Purpose
Step 1	Start Services for Wireless Infrastructure Synchronization, on page 771	In Cisco Unified Serviceability, start services that support the Location Awareness feature.

	Command or Action	Purpose
Step 2	Configure Wireless Access Point Controller, on page 772	Synchronize the database with a Cisco wireless access point controller. The sync imports the wireless infrastructure into the database. **Tip** Set up a sync schedule for automatic updates.
Step 3	Insert Infrastructure Devices, on page 772	Optional. If you want to add your wireless infrastructure from Cisco Prime Infrastructure, or if you are using a third-party wireless LAN controller, use Bulk Administration to update the database from a CSV file. **Note** This method does not allow you to set up automatic updates.
Step 4	Update Infrastructure Devices, on page 774	Optional. If you want to update the existing infrastructure device records, use Bulk Administration to update the database from a CSV file.
Step 5	Deactivate Infrastructure Device from Tracking, on page 775	Optional. If your synchronization includes access points that you do not want to track (for example, if the synchronization pulls in access points from a lab), you can deactivate the access point and Cisco Unified Communications Manager will not track updates to the access point.

Start Services for Wireless Infrastructure Synchronization

Use this procedure to start services that support synchronization with a Cisco Wireless LAN Controller in support of the Location Awareness feature.

Procedure

Step 1 Log in to Cisco Unified Serviceability and choose **Tools** > **Service Activation**.

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

Step 3 Make sure that the following services are checked:

- **Cisco CallManager**
- **Cisco AXL Web Service**
- **Cisco Wireless Controller Synchronization Service**

Step 4 Optional. If you want to use Bulk Administration to import your network infrastructure from a CSV file, make sure that **Bulk Provisioning Service** is checked.

Step 5 Click **Save**.

Configure Wireless Access Point Controller

Use this procedure to synchronize the database with a Cisco wireless access point controller. During the sync, Unified Communications Manager updates its database with the wireless access point infrastructure that the controller manages. You can add up to fifty wireless access point controllers.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Advanced Features** > **Device Location Tracking Services** > **Wireless Access Point Controllers**.

Step 2 Select the controller that you want to configure:

- Click **Find** and select the controller to edit an existing controller.
- Click **Add New** to add a new controller.

Step 3 In the **Name** field, enter the IP address or hostname for the controller.

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

Step 5 Complete the SNMP settings that will be used for SNMP messaging to the controller:

a) From the **SNMP Version** drop-down list, select the SNMP version protocol that the controller uses.
b) Complete the remaining SNMP authentication fields.For more information on the fields and their configuration options, see Online Help.
c) Click the **Test SNMP Settings** to confirm that you entered valid SNMP settings.

Step 6 If you want to configure scheduled syncs to regularly update the database:

a) Check the **Enable scheduled synchronization to discover Infrastructure Devices** check box.
b) In the **Perform a Re-sync Every** fields, create the synchronization schedule.

Step 7 Click **Save**.

Step 8 (Optional) To update the database immediately, click **Synchronize**.

Optional. If the synchronization pulls in access points that you do not want to track (for example, lab equipment or access points that are not in use) you can remove the access point from tracking.

Insert Infrastructure Devices

Use this procedure to complete a bulk import of your wireless Access Point infrastructure from a CSV file into the Unified Communications Manager database. You can use this procedure to import a CSV file that was exported from Cisco Prime Infrastructure or if you want to import access points from a third-party wireless Access Point controller.

Note The Infrastructure Device Utility tool retrieves access points from the Meraki dashboard. The tool formats this data for use on the **Insert Infrastructure Device** page. You can view this utility in the **Upload/Download Files** page.

Before you begin

You must have a data file in comma separated value (CSV) format with the following delineated columns:

- AccessPoint or Switch Name
- IPv4 Address
- IPv6 Address
- BSSID—Required for Wireless Access Protocol (WAP) infrastructure devices.
- Description—A location identifier, a combination of switch type and location, or another meaningful identifier

Note This note applies to releases 12.5(1)SU6, 14SU1, and later releases through 15SU4.

You can define both an IPv4 and IPv6 address, or you can define an IPv4 or an IPv6 address.

For Meraki Access Points, the Unified Communications Manager updates the Basic Service Set Identifiers (BSSID) in the Database after normalizing it to its base BSSID. For more information about BSSID calculation for Meraki Access Points, see Calculating Cisco Meraki BSSID MAC Addresses.

For Non- Meraki Access Points, the Unified CM updates the BSSID in the database by masking the last byte with 0. This masking logic helps Unified CM to uniquely identify the Access Point as opposed to the BSSIDs for the individual channels on the Access Point.

Procedure

Step 1 Choose **Bulk Administration** > **Infrastructure Device** > **Insert Infrastructure Device**.

Step 2 In the **File Name** field, choose the CSV data file that you created for this transaction.

Step 3 In the **Job Information** area, enter the Job description.

The default description is **Insert Infrastructure Device**.

Step 4 Select when you want to run the job:

- Select the **Run Immediately** radio button, if you want to run the job immediately.
- Select the **Run Later** radio button, if you want to schedule the job for later.

Step 5 Click **Submit**.
If you chose to run the job immediately, the job runs.

Step 6 If you chose to run the job later, schedule when the job runs:

a) Choose **Bulk Administration** > **Job Scheduler**.
b) Click **Find** and select the job that you created.
c) In the **Job Scheduler** window, schedule when you want to run the job.
d) Click **Save**.
At the scheduled time, the job runs.

Update Infrastructure Devices

Use this procedure to support updating the existing infrastructure device records using BAT.

Note Location information update is not supported.

Before you begin

- Records must exist in the **Infrastructure Device table** before an update can be performed.
- You must provide at least one of the following: IPv4 address, IPv6 address, or BSSID. All fields cannot be empty, and you may provide more than one.
- To update an access point or a switch, provide the old and new values for one of the following: BSSID, IPv4 address, or IPv6 address. If you are entering BSSID, the BSSID field must contain a single value.
- Two devices cannot have the same BSSID.
- The entered BSSID will not be masked or normalized.

Procedure

Step 1 Choose **Bulk Administration** > **Infrastructure Device** > **Update Infrastructure Device**.

Step 2 In the **File Name** field, choose the CSV data file that you created for this transaction.

Step 3 In the **Job Information** area, enter the Job description.

The default description is **Update Infrastructure Device**.

Step 4 Select when you want to run the job:

- Select the **Run Immediately** radio button, if you want to run the job immediately.
- Select the **Run Later** radio button, if you want to schedule the job for later.

Step 5 Click **Submit**.
If you chose to run the job immediately, the job runs.

Step 6 If you chose to run the job later, schedule when the job runs:

a) Choose **Bulk Administration** > **Job Scheduler**.
b) Click **Find** and select the job that you created.
c) In the **Job Scheduler** window, schedule when you want to run the job.

d) Click **Save**.
At the scheduled time, the job runs.

Deactivate Infrastructure Device from Tracking

If the synchronization includes access points or switches that you do not want to track (for example, if the sync pulls in lab equipment or access points that are not in use), you can deactivate the access point or switch from tracking. Unified Communications Manager will not update the status for the access point or switch.

Procedure

Step 1 In Cisco Unified CM Administration, choose **Advanced Features** > **Device Location Tracking Services** > **Switches and Access Points**.

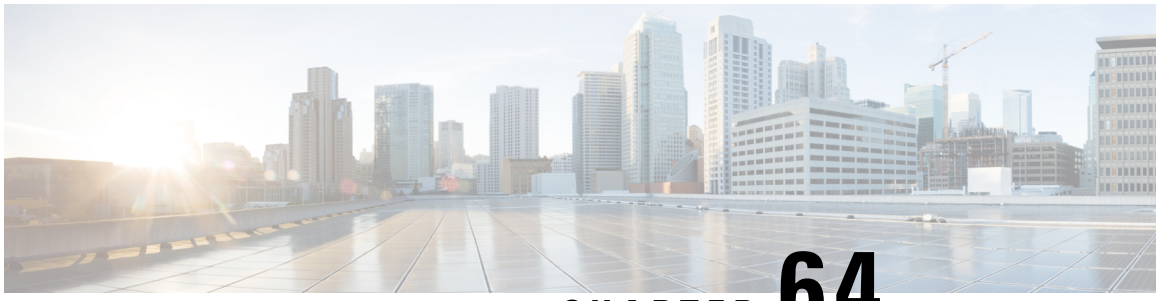
Step 2 Click **Find** and select the switch or access point that you want to stop tracking.

Step 3 Click **Deactivate Selected**.

Related Documentation

After you complete your system configuration, and your system is up and running, you can use tasks in the following chapter to manage your infrastructure on an ongoing basis:

"Manage Infrastructure", Administration Guide for Cisco Unified Communications Manager and IM and Presence Service



CHAPTER 64

Configure Flexible DSCP Marking and Video Promotion

- Flexible DSCP Marking and Video Promotion Overview, on page 777
- Custom QoS Settings for Users, on page 778
- Traffic Class Label, on page 779
- DSCP Settings Configuration Task Flow, on page 779
- Flexible DSCP Marking and Video Promotion Interactions, on page 783
- Flexible DSCP Marking and Video Promotion Restrictions, on page 783

Flexible DSCP Marking and Video Promotion Overview

Devices and applications use Differentiated Services Code Point (DSCP) markings to indicate the Quality of Service (QoS) treatment of IP communications. For example, desktop video endpoints may use multimedia conferencing AF41 marking for video media streams, while high-definition video room systems may use real-time interactive CS4 marking. When an application sends and receives IP communications to and from the same type of application, the DSCP markings are symmetric, and the QoS treatments of the IP communications that each application sends and receives are the same. However, when an application sends and receives media to and from a different type of application, the DSCP markings may be asymmetric, and the QoS treatments of the IP communications that each application sends and receives may be inconsistent. For example, the QoS treatment of the video media stream that a video room system receives from a desktop video endpoint may be inadequate to support the expected quality of the video room system.

Devices and applications are subjected to Call Admission Control (CAC) to ensure that adequate bandwidth is available for the duration of established sessions. The bandwidth that is utilized by established sessions is updated as the sessions begin and end. Attempts to establish new sessions that would exceed the available bandwidth are blocked. The amount of bandwidth available may be tracked independently for devices and applications of different types. For example, independent tracking of bandwidth may be available for desktop video endpoints and high-definition video room systems to send and receive video media streams.

When devices and applications of the same type send and receive communications, the same type of bandwidth deductions are made in each direction. However, when devices and applications of different types send and receive communications, different types of bandwidth deductions must be made in each direction. Moreover, the bandwidth deductions are usually symmetric in amount, by design, to reflect the usual behavior of an IP network. As a result, when devices and applications of different types send and receive communications, the total bandwidth deductions may be up to double the amount of network bandwidth that is actually utilized.

This inconsistency in bandwidth accounting may cause attempts to establish new sessions to be blocked unnecessarily.

The Flexible DSCP Marking and Video Promotion feature allows you to configure a Video Promotion policy that reconciles the inconsistency in bandwidth accounting in favor of the application that receives more favorable CAC and QoS treatment. For example, if a session between a desktop video endpoint and a high-definition video room system is reconciled in favor of the video room system, then the reconciliation is deemed a promotion for the desktop video endpoint.

When reconciliation is in effect between devices and applications of different types, bandwidth is deducted only for the type of application that is favored by reconciliation. If sufficient bandwidth is available for a session of this type to be admitted, the device or application of the type that is not favored by reconciliation is instructed to change the DSCP markings that it uses to those that are used by the device or application of the type that is favored by reconciliation. For example, if a desktop video endpoint is promoted in a session with a high-definition video room system, bandwidth accounting takes place as if the desktop video endpoint were an application of the same type as the video room system. The desktop video endpoint is instructed to change its DSCP markings to those that are used by the video room system. The QoS treatment is consistent in both directions, bandwidth is deducted for a session between devices and applications of the same type as the video room system, and bandwidth is not deducted for a session between devices and applications of the same type as the desktop video endpoint.

When you activate the Flexible DSCP Marking and Video Promotion feature, Unified Communications Manager dynamically signals desktop video devices a Traffic Class Label that is indicative of the DSCP marking for each negotiated media stream.

Custom QoS Settings for Users

You can customize Quality of Service (QoS) settings within a SIP profile and apply those settings to your users. The **SIP Profile Configuration** window has been enhanced with the following types of QoS settings:

- Custom DSCP values for audio and video streams
- Custom UDP port ranges for audio and video streams

Custom DSCP Values for Audio and Video

You can configure DSCP values for audio and video calls within a SIP profile and apply them to the SIP phones that use that profile. The **SIP Profile Configuration** window includes custom DSCP settings for the following types of calls:

- Audio calls
- Video calls
- Audio portion of a video call
- TelePresence calls
- Audio portion of a TelePresence call

If your company has a set of employees, such as a sales force, or a CEO, who require higher QoS priority settings than the majority of your employees, you can use the SIP profile configurations to configure custom DSCP values for those users. The settings within the SIP profile override the corresponding clusterwide service parameter settings.

Custom UDP Port Ranges for Audio and Video

You can configure separate UDP port ranges for the audio stream and video stream of a SIP call. Because video typically requires considerably more bandwidth than audio, creating dedicated port ranges for each media type simplifies network bandwidth management. It also protects against audio stream degradation by guaranteeing that the audio stream will have a dedicated channel that is separate from the higher-bandwidth video stream.

You can apply this configuration by setting the **Media Port Ranges** field in the SIP profile to **Separate Port Ranges for Audio and Video**. You can then apply the configuration to a phone by associating the SIP profile to a phone.

Traffic Class Label

The Flexible DSCP and Video Promotion feature uses the Traffic Class Label (TCL) to instruct the SIP endpoint dynamically to mark its DSCP on a per call basis, based on the Video Promotion policy that you configure. Because TCL is a SIP Session Description Protocol (SDP) attribute that is defined per media line, the TCL and its associated DSCP markings can be different for the audio media line and the video media line of a video call. You can choose different DSCP markings for the audio stream and the video stream of the video call.

DSCP Settings Configuration Task Flow

Perform the following tasks to configure DSCP values and a video promotion policy for your network.

Procedure

	Command or Action	Purpose
Step 1	Configure Flexible DSCP Marking and Video Promotion Policy, on page 779	Configure a video promotion policy to handle the different types of video.
Step 2	Configure Custom QoS Policy for Users, on page 781	If your company has users that require higher priority than other users in your company, configure a SIP Profile that includes custom DSCP values for audio and video streams. For example, if your company has a telephone sales force or CEO whom require higher priority, you can apply the customized SIP profile to those users' phones.

Configure Flexible DSCP Marking and Video Promotion Policy

Follow these steps to configure a video promotion policy to handle the different types of video.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Service Parameters**.

Step 2 From the **Server** drop-down list, choose the server where you want to configure the parameters.

Step 3 From the **Service** drop-down list, choose the **Cisco CallManager (Active)** service.

If the service does not display as active, ensure that the service is activated in Cisco Unified Serviceability.

Step 4 To configure a Video Promotion policy that promotes desktop video endpoints to immersive video endpoints, set the **Use Video BandwidthPool for Immersive Video Calls** parameter to **False** and set the **Video Call QoS Marking Policy** parameter to **Promote to Immersive**.

Step 5 To configure other parameters, scroll to the appropriate area of the **Service Parameter Configuration** window and update the parameter values. See Flexible DSCP Marking and Video Promotion Service Parameters, on page 780 for information about the service parameters and their configuration options.

Step 6 Click **Save**.

Flexible DSCP Marking and Video Promotion Service Parameters

Note For more information about the service parameters, click the parameter name or click the question mark (?) icon that displays in the **Service Parameter Configuration** window.

Table 80: Flexible DSCP Marking and Video Promotion Service Parameters

Parameter	Description
Clusterwide Parameters (System - QoS)	This section of service parameters includes clusterwide DSCP values for a wide range of audio and video call types, including DSCP for audio calls, video calls, the audio portion of a video call, TelePresence calls, and the audio portion of a TelePresence call. It is highly recommended that you keep these parameters set to the default value unless a Cisco support engineer instructs otherwise.
Clusterwide Parameters (Call Admission Control)	
Video Call QoS Marking Policy	This parameter allows you to configure a Promote to Immersive policy that reconciles bandwidth allocation inconsistencies between a desktop video endpoint and a Cisco TelePresence immersive video endpoint in favor of the immersive endpoint. When promotion is performed, the audio and video bandwidth are reserved from the immersive bandwidth pool allocation. The policy of Promote to Immersive takes effect only for calls between an immersive video device and a desktop video device that supports flexible DSCP marking.
Clusterwide Parameters (System - Location and Region)	

Parameter	Description
Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio)	This parameter specifies the default maximum total bit rate for each immersive video call within a particular region, when the **Use System Default** option is selected as the **Max Immersive Video Call Bit Rate** in the **Region Configuration** window for the relationship of the region with itself.
Default Interregion Max Immersive Video Call Bit Rate (Includes Audio)	This parameter specifies the default maximum total bit rate for each immersive video call between a particular region and another region, when the **Use System Default** option is selected as the **Max Immersive Video Call Bit Rate** in the **Region Configuration** window for the relationship of the region with the other region.
Use Video BandwidthPool for Immersive Video Calls	This parameter specifies whether Unified Communications Manager reserves bandwidth from the desktop video bandwidth pool for immersive video calls.

Configure Custom QoS Policy for Users

Perform the following tasks to set up a custom Quality of Service (QoS) policy for users. You may want to apply a custom policy if a set of users within your company has different QoS requirements from the rest of the company such as telephone sales force or a CEO.

Procedure

	Command or Action	Purpose
Step 1	Configure Custom QoS Settings in SIP Profile, on page 781	Configure a SIP Profile with customized DSCP values and a UDP port range for audio and video streams.
Step 2	Apply Custom QoS Policy to a Phone, on page 782	Apply the SIP Profile to a phone. The DSCP settings in the SIP Profile override the DSCP clusterwide service parameter settings..

Configure Custom QoS Settings in SIP Profile

Configure custom DSCP values and UDP port ranges for the phones that use this SIP Profile. You can use these settings to configure a customized QoS policy that you can apply to specific phones and users within your network. You may want to do this if you want to apply specific QoS settings to specific users within your enterprise, such as a sales force, or a CEO.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.

Step 2 Perform either of the following steps:

- Click **Find** and select an existing SIP Profile.

- Click **Add New** to create a new SIP Profile.

Step 3 From the **Media Port Ranges** field, select whether you want to assign a single UDP port range that handles both audio and video media, or separate port ranges for audio and video streams.

- If you want to configure a single port range for audio and video media, enter the range of ports in the **Start Media Port** and **Stop Media Port** fields. The possible port values are between 2048 and 65535.
- If you want separate port ranges for audio and video streams, enter the range of audio ports using the **Start Audio Port** and **Stop Audio Port** fields. Enter the range of video ports using the **Start Video Port** and **Stop Video Port** fields. The possible port values for each are between 2048 and 65535.The two port ranges must not overlap.

Step 4 In the following fields, configure customized DSCP values for audio and video streams.

- DSCP for Audio Calls
- DSCP for Video Calls
- DSCP for Audio Portion of Video Calls
- DSCP for TelePresence Calls
- DSCP for Audio Portion of TelePresence Calls

Note
By default, each of the above fields is configured to use the value from a corresponding service parameter. If you assign new values, the new value overrides the service parameter setting.

Step 5 Complete the remaining fields in the **SIP Profile Configuration** window. For help with the fields and their settings, refer to the online help.

Step 6 Click **Save**.

Apply Custom QoS Policy to a Phone

Use this procedure to apply a SIP Profile that contains customized QoS settings, including DSCP values and a UDP port range for audio and video media. When you apply this SIP profile to a phone, the phone uses the custom settings from the SIP Profile.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.

Step 2 Perform any one of the following steps:

- Click **Find** and select an existing phone.
- Click **Add New** to create a new phone.

Step 3 From the **SIP Profile** drop-down list, select the SIP profile that you set up with the custom DSCP values and UDP port range values.

Step 4 Complete the remaining fields in the **Phone Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 5 Click **Save**.

Flexible DSCP Marking and Video Promotion Interactions

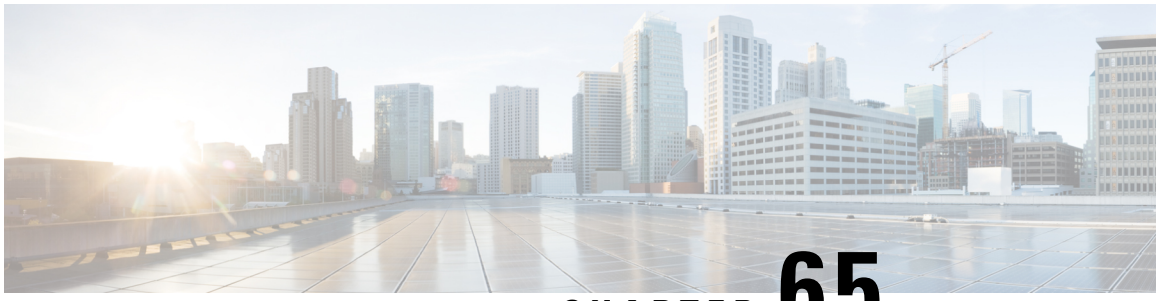
Table 81: Flexible DSCP Marking and Video Promotion Interactions

Device	Interaction
SIP Intercluster Trunks	The Flexible DSCP Marking and Video Promotion feature is supported over SIP intercluster trunks.
Skinny Client Control Protocol (SCCP) Devices	The Flexible DSCP Marking and Video Promotion feature is supported for SCCP devices.
Pass-Through MTPs	If pass-through MTPs are inserted in a call, Unified Communications Manager signals the MTP to mark the packets with the DSCP marking that is expected from the endpoint device that originally emitted the packet for the video stream. If the two endpoints on a call use different DSCP markings (for example, a Cisco TelePresence immersive video endpoint and a desktop video endpoint without Video Promotion), the MTPs preserve the DSCP marking in each stream direction.

Flexible DSCP Marking and Video Promotion Restrictions

Table 82: Flexible DSCP Marking and Video Promotion Restrictions

Restriction	Description
Trunks and gateways	The Flexible DSCP Marking and Video Promotion feature is not supported over H.323 trunks and Media Gateway Control Protocol (MGCP) gateways.
Multilevel Precedence and Preemption	Cisco recommends that you do not use the Flexible DSCP Marking and Video Promotion feature with Multilevel Precedence and Preemption (MLPP) service calls. When you need MLPP service functionality, Cisco recommends that you set the Video Call QoS Marking Policy and Use Video BandwidthPool for Immersive Video Calls service parameters to their default values. With default values for the Video Call QoS Marking Policy and Use Video BandwidthPool for Immersive Video Calls service parameters, Unified Communications Manager and endpoints use MLPP DSCP markings for the media packets.
SIP video endpoints	The Flexible DSCP Marking and Video Promotion feature is dependent on desktop SIP video endpoint support. Currently, only Cisco DX650 series SIP phones provide the required endpoint support.



CHAPTER 65

Separate Calling Party Number and Billing Number in SIP

- External Presentation Name and Number Overview, on page 785
- Call Processing, on page 786
- Directory Number Overview, on page 788
- SIP Profile Overview, on page 792
- SIP Trunk Overview, on page 794
- Intercluster SME Call Flows, on page 799

External Presentation Name and Number Overview

Cisco Unified Communications Manager Administration can be configured to contain separate calling party and presentation number.

In previous releases, Cisco Unified Communications Manager cannot be configured on a per line basis to have a different number sent to the PSTN in the FROM and PAID header. If a group of users is configured to present the same Calling Line Identification Number to PSTN users, which is a nongeographic E.164 number and cannot be used for billing. Therefore, the users actual DDI must be sent in a different field than the presentation number. With this release, Cisco Unified Communications Manager supports External Presentation Name and Number that is different from existing Identification Name and Number. The configured Presentation Name and Number are for display purpose on the following devices:

- SIP
- SCCP
- Single Number Reach Destination (SNRD)
- CTIRD
- SparkRD

Configuration Overview

You can configure the external presentation name and number feature on the following pages:

- Directory Number Configuration

- SIP Profile Configuration
- Trunk Configuration

Note

- When you configure the External Presentation Information on the SIP Profile Configuration page, the value of **External Presentation Number** and **External Presentation Name** on the SIP Profile Configuration page is used, overriding the settings configured on the Directory Number page.
- When you configure the Presentation Information on the Trunk Configuration page, the value of **Presentation Number** and **Presentation Name** on the Trunk Configuration page is used, overriding the settings configured on the SIP Profile Configuration and Directory Number Configuration pages.

Call Processing

This section describes the incoming and outgoing call behavior when you configure the external presentation name and number feature.

Incoming Call Process

Cisco Unified Communications Manager looks for FROM and PAID header information when there is a call initiated from the PSTN network. The FROM header contains the external presentation name and number (if configured). However, this is not the real identity of a user, it is used only for display purpose. The PAID header contains the identity (original DN or DDI) of the user.

If FROM and PAID headers have different numbers and **Enable External Presentation Name and Number** option is enabled in the SIP Profile Configuration page and **Display External Presentation Name and Number** service parameter value is set to **`True`**, then Cisco Unified Communications Manager displays the FROM header information (configured external presentation name and number) on the called device. Similarly, if an option is disabled, Cisco Unified Communications Manager displays PAID header information (user's original DN or DDI) on the called device.

Note

- By default, **Enable External Presentation Name and Number** field is unchecked.
- Default value of a service parameter **Display External Presentation Name and Number** is False.

Invite Received from the PSTN Network

```
From: "Customer Care" <sip:1800000@example.com>;
To: <sip:someone@example.com>
P-Asserted-Identity: "Your personal adviser <sip:user1@example.com>
Remote-Party-ID: "Your personal adviser <sip:user1@example.com>
```

In the preceding example, FROM header contains a number different from the PAID header. If you check **Enable External Presentation Name and Number** check box and set **Display External Presentation Name and Number** value to **`True`**, Cisco Unified Communications Manager displays **`Customer Care / 1800000`** on the called device.

If you uncheck the **Enable External Presentation Name and Number** check box or set the **Display External Presentation Name and Number** to **`False`**, then Cisco Unified Communications Manager displays **`Your personal adviser / user1@example.com`** on the called device.

Outgoing Call Process

Let us assume, a user configured with External Presentation Name and Number initiates a call to a PSTN network through the SIP trunk with **Enable External Presentation Name and Number** configured in its SIP Profile. Then, Cisco Unified Communications Manager sends the configured External Presentation Information in the FROM header of the outgoing SIP message and displays on the called device.

If **Enable External Presentation Name and Number** option is disabled or **External Presentation Number** and **External Presentation Name** fields are not configured, the Cisco Unified Communications Manager sends the directory number information in the FROM and PAID headers and displays on the called device.

External Presentation Number Mask Operation

Cisco Unified Communications Manager allows you to mask the external presentation number, to be displayed on the called device. You can mask the presentation number on the Directory Number Configuration, SIP Profile Configuration, and Trunk Configuration pages.

When you enter the digits in the **External Presentation Number** field with trailing X, the value of X is replaced with the directory number information starting from right to left.

Mask Operation on Directory Number Configuration

If you mask **External Presentation Number** as 180011XXXX on Directory Number Configuration page for a Directory Number 5551234, then Cisco Unified Communications Manager displays the presentation number as `1800111234` on the called device.

Mask Operation on SIP Profile Configuration

Let us assume **External Presentation Number** on Directory Number page is 180011XXXX and if you mask **External Presentation Number** on SIP Profile Configuration page as 180022XXXX for a Directory Number 5551234, then Cisco Unified Communications Manager displays the presentation number as `1800221234` on the called device.

Mask Operation on Trunk Configuration

Let us assume **External Presentation Number** on Directory Number page is 180011XXXX and on SIP Profile Configuration page is 180022XXXX. If you mask **Presentation Number** on Trunk Configuration page as 180033XXXX for a Directory Number 5551234, then Cisco Unified Communications Manager displays the presentation number as `1800331234` on the called device.

Directory Number Overview

In Cisco Unified Communications Manager Administration, use the **Call Routing** > **Directory Number** menu path to configure Directory Numbers (DNs). Using Cisco Unified Communications Manager Administration, you can configure and modify the DNs that are assigned to specific phones.

A new section **External Presentation Information** is added on the Directory Number Configuration page. The administrator can now configure the presentation name and number of their choice to display on the supported devices for external calls. If an administrator does not want to show users' identity, they have a privilege to display configured **External Presentation Number** and **External Presentation Name** as Anonymous on the called party device.

Directory Number Configuration Tasks

Procedure

	Command or Action	Purpose
Step 1	Add a new end user using one of the following methods: • Import an End User from LDAP, on page 788 • Add an End User Manually, on page 789	If your system is synchronized with a company LDAP directory, you can import the new end user directly from LDAP. Otherwise, you can add and configure the end user manually.
Step 2	Assign a phone to new or existing end user by performing one of the following tasks: • Add New Phone for End User , on page 790 • Move an Existing Phone to a End User, on page 791	You can use the 'Add New Phone' procedure to configure a new phone for the end user using settings from a universal device template. You can also use the 'Move' procedure to assign an existing phone already configured or pre-configured.
Step 3	Configure External Presentation Information on DN, on page 791	To configure the external presentation number and external presentaion name for DNs that are assigned to specific phones.

Import an End User from LDAP

Perform the following procedure to manually import a new end user from a company LDAP directory. If your LDAP synchronization configuration includes a feature group template with a user profile that includes universal line and device templates and a DN pool, the import process automatically configures the end user and primary extension.

Note You cannot add new configurations (for example, adding a feature group template) into an LDAP directory sync after the initial sync has occurred. If you want to edit an existing LDAP sync, you must either use Bulk Administration, or configure a new LDAP sync.

Before you begin

Before you begin this procedure make sure that you have already synchronized Cisco Unified Communications Manager with a company LDAP directory. The LDAP synchronization must include a feature group template with universal line and device templates.

Procedure

Step 1 In Cisco Unified CM Administration, choose **System** > **LDAP** > **LDAP Directory**.

Step 2 Click **Find** and select the LDAP directory to which the user is added.

Step 3 Click **Perform Full Sync**.
Cisco Unified Communications Manager synchronizes with the external LDAP directory. Any new end users in the LDAP directory are imported into the Cisco Unified Communications Manager database.

What to do next

If the user is enabled for self-provisioning, the end user can use the Self-Provisioning Interactive Voice Response (IVR) to provision a new phone. Otherwise, perform one of the following tasks to assign a phone to the end user:

- Add New Phone for End User , on page 790
- Move an Existing Phone to a End User, on page 791

Add an End User Manually

Perform the following procedure to add new end user and configure them with an access control group and a primary line extension.

Note Make sure that you have already set up an access control groups that has the role permissions to which you want to assign your user. For details, see the "Manage User Access" chapter.

Before you begin

Verify that you have a user profile configured that includes a universal line template. If you need to configure a new extension, Cisco Unified Communications Manager uses the settings from the universal line template to configure the primary extension.

Procedure

Step 1 In Cisco Unified CM Administration, choose **User Management** > **User/Phone Add** > **Quick User/Phone Add**.

Step 2 Enter the **User ID** and **Last Name**.

Step 3 From the **Feature Group Template** drop-down list, select a feature group template.

Step 4 Click **Save**.

Step 5 From the **User Profile** drop-down list, verify that the selected user profile includes a universal line template.

Step 6 From the **Access Control Group Membership** section, click the + icon.

Step 7 From the **User is a member of** drop-down list, select an access control group.

Step 8 Under **Primary Extension**, click the + icon.

Step 9 From the **Extension** drop-down list, select a DN that displays as **(available)**.

Step 10 If all line extensions display as **(used)**, perform the following steps:

a) Click the **New...** button.
The **Add New Extension** popup displays.
b) In the **Directory Number** field, enter a new line extension.
c) From the **Line Template** drop-down list, select a universal line template.
d) Click **OK**.
Cisco Unified Communications Manager configures the directory number with the settings from the universal line template.

Step 11 (Optional) Complete any additional fields in the **Quick User/Phone Add Configuration** window.

Step 12 Click **Save**.

What to do next

Perform one of the following procedures to assign a phone to this end user:

- Add New Phone for End User , on page 790
- Move an Existing Phone to a End User, on page 791

Add New Phone for End User

Perform the following procedure to add a new phone for a new or existing end user. Make sure that the user profile for the end user includes a universal device template. Cisco Unified Communications Manager uses the universal device template settings to configure the phone.

Before you begin

Perform one of the following procedures to add an end user:

- Add an End User Manually, on page 789
- Import an End User from LDAP, on page 788

Procedure

Step 1 In Cisco Unified CM Administration, choose **User Management** > **User/Phone Add** > **Quick/User Phone Add**.

Step 2 Click **Find** and select the end user for whom you want to add a new phone.

Step 3 Click the **Manage Devices**.
The Manage Devices window appears.

Step 4 Click **Add New Phone**.
The Add Phone to User popup displays.

Step 5 From the **Product Type** drop-down list, select the phone model.

Step 6 From the **Device Protocol** drop-down list select SIP or SCCP as the protocol.

Step 7 In the **Device Name** text box, enter the device MAC address.

Step 8 From the **Universal Device Template** drop-down list, select a universal device template.

Step 9 If the phone supports expansion modules, enter the number of expansion modules that you want to deploy.

Step 10 If you want to use Extension Mobility to access the phone, check the **In Extension Mobility** check box.

Step 11 Click **Add Phone**.
The Add New Phone popup closes. Cisco Unified Communications Manager adds the phone to the user and uses the universal device template to configure the phone.

Step 12 If you want to make additional edits to the phone configuration, click the corresponding Pencil icon to open the phone in the **Phone Configuration** window.

Move an Existing Phone to a End User

Perform this procedure to move an existing phone to a new or existing end user.

Procedure

Step 1 In Cisco Unified CM Administration, choose **User Management** > **User/Phone Add** > **Quick/User Phone Add**.

Step 2 Click **Find** and select the user to whom you want to move an existing phone.

Step 3 Click the **Manage Devices** button.

Step 4 Click the **Find a Phone to Move To This User** button.

Step 5 Select the phone that you want to move to this user.

Step 6 Click **Move Selected**.

Configure External Presentation Information on DN

Perform the following procedure to configure the external presentation information for DNs that are assigned to specific phones.

Before you begin

- Check the **Enable External Presentation Name and Number** check box on the SIP Profile Configuration page.
- Perform one of the following procedures to add an end user:
  - Add an End User Manually, on page 789
  - Import an End User from LDAP, on page 788
- Assign a phone to a new or existing end user by performing one of the following tasks:

- Add New Phone for End User , on page 790
- Move an Existing Phone to a End User, on page 791

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Directory Number**.

Step 2 From the **Find and List Directory Numbers** page, perform one of the following steps:

- To update a DN, click **Find** and select the Directory Number for which you want to display unique identity.
- To create a new Directory Number, click **Add New**.

Step 3 In the **External Presentation Information** section, enter the name and number that you want to display on the called device.

Note
- **External Presentation Number** field accepts up to 32 digits and can contain the following characters: [0-9, X, *, #, \, +].
- **External Presentation Name** field accepts up to 50 characters.

Step 4 (Optional), if you want to show configured **External Presentation Number** and **External Presentation Name** as anonymous, check the **Anonymous External Presentation** check box.

Note
- By default, the **Anonymous External Presentation** field is unchecked.
- If you check the **Anonymous External Presentation** field:

  External Presentation Number and **External Presentation Name** fields are noneditable. Also, the entries from these fields disappear.

Step 5 Complete the remaining fields on the **Directory Number Configuration** page. For help with the fields and their settings, refer to the online help.

Step 6 Click **Save**.

SIP Profile Overview

A SIP profile is a template that comprises common SIP settings. You must assign a SIP profile for every SIP trunk and SIP device in your network. When you configure a SIP profile and then assign that profile to a SIP trunk, or a SIP device, the system applies the configured SIP settings to that trunk or device.

SIP Profile Configuration Tasks

Procedure

	Command or Action	Purpose
Step 1	Configure SIP Profiles, on page 793	Use this procedure to configure a SIP profile.
Step 2	Configure External Presentation Information on SIP Profile, on page 793	To configure the external presentation number and external presentation name for a SIP profile.

Configure SIP Profiles

Use this procedure to configure a SIP profile with common SIP settings that you can assign to SIP devices and trunks that use this profile.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.

Step 2 Perform one of the following steps:

- Click **Find** and select the SIP profile to edit an existing profile, .
- Click **Add New** to create a new profile.

Step 3 If you want your SIP phones and trunks to support IPv4 and IPv6 stacks, check the **Enable ANAT** check box.

Step 4 If you want to assign an SDP transparency profile to resolve SDP interoperability, from the **SDP Transparency Profile** drop-down list.

Step 5 If you want to assign a normalization or transparency script to resolve SIP interoperability issues, from the **Normalization Script** drop-down list, select the script.

Step 6 (Optional) Check the **Send ILS Learned Destination Route String** check box for Global Dial Plan Replication deployments where you may need to route calls across a Cisco Unified Border Element.

Step 7 Complete the remaining fields in the **SIP Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 8 Click **Save**.

Configure External Presentation Information on SIP Profile

Use this procedure to configure the separate external presentation name and number on SIP Profile Configuration page.

Before you begin

- Check the **Enable External Presentation Name and Number** check box on the **SIP Profile Configuration** page.
- Set **Display External Presentation Name and Number** parameter value to `True` under Clusterwide Parameters (Device-Phone) section on **Service Parameter Configuration** page.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.

Step 2 Perform one of the following steps:

- To edit an existing profile, click **Find** and select the SIP profile.
- To create a new profile, click **Add New**.

Step 3 In the **External Presentation Information** section, enter the name and number that you want to display on the called device.

Note

- **External Presentation Number** field accepts up to 32 digits and can contain the following characters: [0-9, X, *, #, \, +].
- **External Presentation Name** field accepts a maximum of 50 characters.

Step 4 (Optional), if you want to show configured **External Presentation Number** and **External Presentation Name** as anonymous, check the **Anonymous External Presentation** check box.

Note

- By default, the **Anonymous External Presentation** field is unchecked.
- If you check the **Anonymous External Presentation** field:

  External Presentation Number and **External Presentation Name** fields are noneditable. Also, the entries from these fields disappear.

Step 5 Complete the remaining fields in the **SIP Profile Configuration** page. For more information on the fields and their configuration options, see the system Online Help.

Step 6 Click **Save**.

SIP Trunk Overview

If you are deploying SIP for call control signaling, configure SIP trunks that connect Cisco Unified Communications Manager to external devices such as SIP gateways, SIP Proxy Servers, Unified Communications applications, remote clusters, or a Session Management Edition.

Within the Cisco Unified CM Administration, the SIP Trunk Configuration window contains the SIP signaling configurations that Cisco Unified Communications Manager uses to manage SIP calls.

SIP Trunk supports the separate presentation name and number that is different from existing Caller ID DN and Caller Name. A new check box **Anonymous Presentation** is provided to show the configured presentation name and number as Anonymous on the called device.

Trunk Configuration Tasks

Procedure

	Command or Action	Purpose
Step 1	Configure SIP Trunk Security Profile, on page 795	Configure SIP trunk security profiles with any security settings that you want to apply to your SIP trunks. For example, you can configure digest authentication, device security mode, and TLS encryption for SIP signaling. If you don't configure SIP trunk security profiles, by default, Cisco Unified Communications Manager applies a nonsecure sip trunk security profile.
Step 2	Configure Common Device Configuration, on page 796	Set up a Common Device Configuration for the trunk. For dual-stack trunks, configure the IP addressing preference.
Step 3	Configure SIP Trunks, on page 797	Configure the SIP trunks in your network. In the **Trunk Configuration** window, configure the SIP settings for your trunks. Assign a SIP profile, SIP trunk security profile, and a Common Device Configuration to your SIP trunk. In addition, assign any SIP normalization or transparency scripts that your trunk connection requires. For example, if your SIP trunk connects to a Cisco TelePresence VCS, you must assign the *vcs-interop* script to the SIP trunk.
Step 4	Configure Presentation Information on SIP Trunks, on page 798	To configure the presentation name and number on SIP Trunk page.

Configure SIP Trunk Security Profile

Configure a SIP Trunk Security Profile with security settings such as digest authentication or TLS signaling encryption. When you assign the profile to a SIP trunk, the trunk takes on the settings of the security profile.

Note If you don't assign a SIP trunk security profile to your SIP trunks, Cisco Unified Communications Manager assigns a nonsecure profile by default.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Security** > **SIP Trunk Security Profile**.

Step 2 Click **Add New**.

Step 3 To enable SIP signaling encryption with TLS, perform the following:

a) From the **Device Security Mode** drop-down list, select **Encrypted**.
b) From the **Incoming Transport Type** and **Outgoing Transport Type** drop-down lists, choose **TLS**.
c) For device authentication, in the **X.509 Subject Name** field, enter the subject name of the X.509 certificate.
d) In the **Incoming Port** field, enter the port on which you want to receive TLS requests. The default for TLS is 5061.

Step 4 To enable digest authentication, do the following

a) Check the **Enable Digest Authentication** check box
b) Enter a **Nonce Validity Timer** value to indicate the number of seconds that must pass before the system generates a new nonce. The default is 600 (10 minutes).
c) To enable digest authentication for applications, check the **Enable Application Level Authorization** check box.

Step 5 Complete the additional fields in the **SIP Trunk Security Profile Configuration** window.For more information on the fields and their configuration options, see Online Help.

Step 6 Click **Save**.

Note
You must assign the profile to a trunk in the **Trunk Configuration** window so that the trunk can uses the settings.

Configure Common Device Configuration

A common device configuration comprises a set of optional set of user-specific feature attributes. If you are deploying IPv6, you can use this configuration to assign IPv6 preferences for SIP trunks or SCCP phones.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Common Device Configuration**.

Step 2 Click **Add New**.

Step 3 For SIP trunks, SIP Phones or SCCP phones, choose a value for the **IP Addressing Mode** drop-down list:

- **IPv4 Only**—The device uses only an IPv4 address for media and signaling.
- **IPv6 Only**—The device uses only an IPv6 address for media and signaling.
- **IPv4 and IPv6 (Default)**—The device is a dual-stack device and uses whichever IP address type is available. If both IP address types are configured on the device, for signaling the device uses the **IP Addressing Mode Preference for Signaling** setting and for media the device uses the **IP Addressing Mode Preference for Media** enterprise parameter setting.

Step 4 If you configure IPv6 in your previous step, then configure an IP addressing preference for the **IP Addressing Mode for Signaling** drop-down list:

- **IPv4**—The dual stack device prefers IPv4 address for signaling.
- **IPv6**—The dual stack device prefers IPv6 address for signaling.
- **Use System Default**—The device uses the setting for the **IP Addressing Mode Preference for Signaling** enterprise parameter.

Step 5 Configure the remaining fields in the **Common Device Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

Step 6 Click **Save**.

Configure SIP Trunks

Use this procedure to configure a SIP trunk. You can assign up to 16 destination addresses for a SIP trunk.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Trunk**.

Step 2 Click **Add New**.

Step 3 From the **Trunk Type** drop-down list, choose **SIP Trunk**.

Step 4 From the **Protocol Type** drop-down list, choose the type of SIP trunk that matches your deployment and click **Next**:

- **None (Default)**
- **Call Control Discovery**
- **Extension Mobility Cross Cluster**
- **Cisco Intercompany Media Engine**
- **IP Multimedia System Service Control**

Step 5 (Optional) If you want to apply a **Common Device Configuration** to this trunk, select the configuration from the drop-down list.

Step 6 Check the **SRTP Allowed** check box if you want to allow encypted media over the trunk.

Step 7 Check the **Run on All Active Unified CM Nodes** check box if you want to enable the trunk for all cluster nodes.

Step 8 Configure the destination address for the SIP trunk:

a) In the **Destination Address** text box, enter an IPv4 address, fully qualified domain name, or DNS SRV record for the server or endpoint that you want to connect to the trunk.

b) If the trunk is a dual stack trunk, in the **Destination Address IPv6** text box, enter an IPv6 address, fully qualified domain name, or DNS SRV record for the server or endpoint that you want to connect to the trunk.

c) If the destination is a DNS SRV record, check the **Destination Address is an SRV** check box.

d) To add additional destinations, click the **(+)**.

Step 9 From the **SIP Trunk Security Profile** drop-down, assign a security profile. If you don't select this option, a nonsecure profile will be assigned.

Step 10 From the **SIP Profile** drop-down list, assign a SIP profile.

Step 11 (Optional) If you want to assign a normalization script to this SIP trunk, from the **Normalization Script** drop-down list, select the script that you want to assign.

Step 12 Configure any additional fields in the **Trunk Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 13 Click **Save**.

Configure Presentation Information on SIP Trunks

Use this procedure to configure the presentation name and number on SIP Trunk page.

Before you begin

- Check the **Enable External Presentation Name and Number** check box on the SIP Profile Configuration page.
- Configure SIP Trunks, on page 797

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Trunk**.

Step 2 Click **Add New**.

Step 3 From the **Trunk Type** drop-down list, choose SIP Trunk.

Step 4 From the **Trunk Service Type** drop-down list, choose the type of SIP trunk that you want to configure:

- **None (Default):** The trunk will not be used for Call Control Discovery, Extension Mobility Cross-Cluster, Intercompany Media Engine, or IP Multimedia System Service Control.
- **Call Control Discovery:** The trunk supports the Call Control Discovery feature.
- **Extension Mobility Cross Cluster:** The trunk supports Extension Mobility Cross Cluster.
- **Cisco Intercompany Media Engine:** The trunk supports the Intercompany Media Engine (IME). Make sure that the IME server is installed before you configure this type of trunk.
- **IP Multimedia System Service Control:** Choose this option to enable the trunk with support for IP Multimedia System Service Control.

Step 5 Click **Next**.

Step 6 In the **Presentation Information** section, enter the name and number that you want to display on the called device.

Note

- **Presentation Number** field accepts up to 32 digits and can contain the following characters: [0-9, X, *, #, \, +].
- **Presentation Name** field accepts a maximum of 50 characters.

Step 7 (Optional) If you want to show the presentation name and number as anonymous, check the **Anonymous Presentation** check box.

Note

- By default, the **Anonymous Presentation** field is unchecked.
- If you check the Anonymous External Presentation field:

  Presentation Number and **Presentation Name** fields are noneditable. Also, the entries from these fields disappear.

Step 8 (Optional) Check the **Send Presentation Name and Number only in the FROM header and not in the other identity headers** check box, if you want to send presentation information that is configured on the SIP Trunk only to FROM header.

Step 9 Configure any additional fields in the **Trunk Configuration** window.For more information on the fields and their configuration options, see Online Help.

Step 10 Click **Save**.

Intercluster SME Call Flows

The Cisco Unified Communications Manager Session Management Edition software is same as the Cisco Unified Communications Manager used mainly for a call routing between clusters or various devices. With this release, Cisco Unified Communications Manager supports the intercluster SME calls.

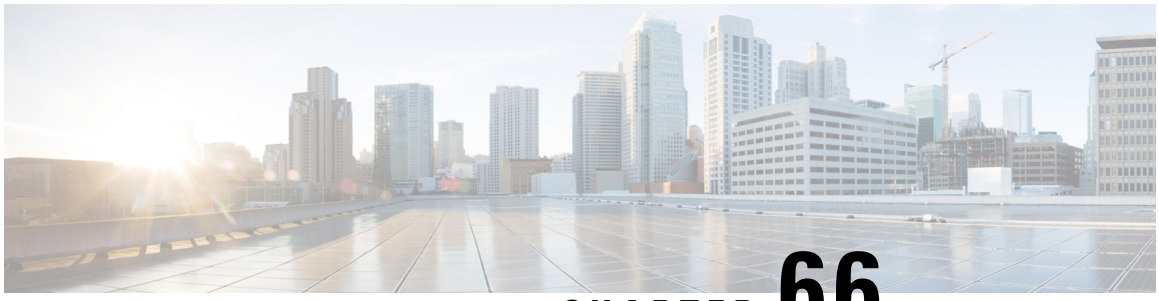
Incoming Calls

Let us assume a user from a PSTN network initiates a call with **Enable External Presentation Name and Number** enabled in its SIP profile. If **Display External Presentation Name and Number** service parameter is set to **`True`**, then Cisco Unified Communications Manager sends the presentation number information to the X-Cisco-Presentation header and displays on the called device. The FROM and PAID headers contains the identity of the user that is the user's DN or DDI.

If **Display External Presentation Name and Number** service parameter is set to **`False`**, then Cisco Unified Communications Manager sends the presentation number information to the X-Cisco-Presentation header. The FROM and PAID headers contains the user’s DN or DDI and displays on the called device.

Outgoing Calls

A user who is configured with **External Presentation Name** and **External Presentation Number** initiates a call to a PSTN network through intercluster SIP trunks. If **Enable External Presentation Name and Number** check box is disabled in its SIP profile, then, Cisco Unified Communications Manager sends the original directory number information in the FROM and PAID headers and displays on the called device and configured External Presentation Information in the X-Cisco-Presentation header. Similarly, if **Enable External Presentation Name and Number** check box is enabled in its SIP profile, Cisco Unified Communications Manager sends the configured External Presentation Information in the FROM header and displays on the called device and original Directory Number in the PAID header.



CHAPTER 66

SIP OAuth Mode

- SIP OAuth Mode Overview, on page 801
- SIP OAuth Mode Prerequisites, on page 802
- SIP OAuth Mode Configuration Task Flow, on page 803

SIP OAuth Mode Overview

Secure registrations to Unified Communications Manager involves a process of updating CTL files, setting up a mutual certificate trust store and so on. If devices are switching between on-premises and off-premises, it is difficult to update LSCs and renew Certificate Authority Proxy Function (CAPF) enrolment each time when a secure registration is completed.

SIP OAuth mode allows you to use OAuth refresh tokens for all devices authentication in secure environments. This feature enhances the security of Unified Communications Manager.

Unified Communications Manager verifies the token presented by the endpoints and serves the configuration files only to authorized ones. OAuth token validation during SIP registration is completed when OAuth based authorization is enabled on Unified Communications Manager cluster and other Cisco devices.

OAuth support for SIP registrations is extended for

- Cisco Jabber devices from Unified Communications Manager 12.5 Release onwards
- SIP Phones from Unified Communications Manager Release 14 onwards

Note By default, TFTP is secure for SIP phones when SIP OAuth is enabled. TFTP file download happens through secured channel, and only for authenticated phones. SIP OAuth provides end to end secure signaling and media encryption without CAPF on-premises as well as over MRA.

The following are the Phone Security Profile Types that can be configured for OAuth:

- Cisco Dual Mode For iPhone (TCT device)
- Cisco Dual Mode For Android (BOT device)
- Cisco Unified Client Service Framework (CSF device)
- Cisco Jabber for Tablet (TAB device)

- Universal Device Template
- Cisco 7811
- Cisco 7821
- Cisco 7832
- Cisco 7841
- Cisco 7861
- Cisco 8811
- Cisco 8832
- Cisco 8832NR
- Cisco 8841
- Cisco 8845
- Cisco 8851
- Cisco 8851NR
- Cisco 8861
- Cisco 8865
- Cisco 8865NR
- Cisco 8875
- Cisco 8875NR
- Cisco 9811
- Cisco 9841
- Cisco 9851
- Cisco 9861
- Cisco 9861NR
- Cisco 9871
- Cisco 9871NR

SIP OAuth Mode Prerequisites

This feature assumes that you have already completed the following:

- Ensure that Mobile and Remote Access is configured and the connection is established between Unified Communication Manager and Expressway. This rule isn't applicable to On-Prem SIP OAuth deployments.
- Ensure that Unified Communications Manager is registered to a Smart or Virtual account with **allow export-controlled** functionality.

- Ensure that the client firmware supports SIP OAuth.
- The phones must trust both the Tomcat and Tomcat-EC certificates for SIP OAuth to work.
- You can generate and download the self-signed Tomcat and Tomcat-EC certificates, or CA signed root certificate, and then upload this certificate as the Phone-Edge-Trust certificates on the Unified Communications Manager system. The IP Phones can accept a maximum of 16 Phone-Edge-Trust certificates. After uploading, the CA signed root certificates are placed into the caconfig.json file. For verification, you can access the URL at: `http://<cucm>:6970/caconfig.json.sgn`. Your deployment must have the following: Unified CM version 14 and above, Cisco IP Phones with SIP Firmware Release 14.0 and above, and Cisco Expressway X12.7 and above (in case of MRA deployments).

SIP OAuth Mode Configuration Task Flow

Complete the following tasks to configure SIP OAuth for your system.

Procedure

	Command or Action	Purpose
Step 1	Upload CA Certificate to the Phone Edge Trust	Upload CA Certificate to the phone edge trust to get the tokens. This step is not applicable for Cisco Jabber device.
Step 2	Enable OAuth Access Token for Devices	**Important** This step is applicable from Release 14 onwards. Enable OAuth for SIP registrations in Cisco IP Phone 7800 and 8800 enterprise series. This step is not applicable for Cisco Jabber device.
Step 3	Configure Refresh Logins, on page 805	Enable oauth with refresh login flow on Unified Communications Manager to register the device via SIP OAuth.
Step 4	Configure OAuth Ports, on page 805	Assign the ports for OAuth for each node that has OAuth registration.
Step 5	Configure OAuth Connection to Expressway-C, on page 806	Configure a mutually authenticated TLS connection to Expressway-C.
Step 6	Enable SIP OAuth Mode, on page 807	Enable OAuth services using a CLI command on the publisher node.
Step 7	Restart Cisco CallManager Service, on page 807	Restart this service on all nodes that have OAuth registrations.
Step 8	Configure Device Security Mode in Phone Security Profile	Configure OAuth support within a Phone Security Profile if you are deploying encryption for the endpoints.

	Command or Action	Purpose
Step 9	(Optional) Configure SIP OAuth Registered Phones for MRA Mode	**Important** This step is applicable from Release 14 onwards. Configure SIP OAuth registered phones in MRA mode. This step is not applicable for Cisco Jabber device.

Upload CA Certificate to the Phone Edge Trust

Use this procedure to upload the root certificate of Tomcat signed certificate to the Phone Edge Trust from the publisher node. The certificate is only visible on the publisher node.

Note This procedure is performed only for Cisco Phones and not applicable for Cisco Jabber.

Procedure

Step 1 From Cisco Unified OS Administration, choose **Security** > **Certificate Management**.

Step 2 Click **Upload Certificate/Certificate chain**.

Step 3 In the **Upload Certificate/Certificate chain** window, from the **Certificate Purpose** drop-down list choose **Phone-Edge-Trust**.

Step 4 In the **Upload File** field, click **Browse** and upload the certificate.

Step 5 Click **Upload**.

Enable OAuth Access Token for Devices

Important This section is applicable from Release 14 onwards.

Use this procedure to enable OAuth access token for phones.

Note Configure this enterprise parameter only for OAuth support for SIP registrations for phones.

The phone certificate (MIC or LSC) must be valid for SIP OAuth to work.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Enterprise Parameters**.

Step 2 In **SSO and OAuth Configuration** section, ensure that the value of **OAuth Access Token for Devices** drop-down list is set to **Implicit:Already registered devices**.

Note
Set the value of **OAuth Access Token for Devices** to **Explicit:Activation Code device onboarding required** to disable implicitly receiving tokens for SIP OAuth registration and only support receiving tokens through activation code. The tokens can then be used for SIP OAuth registration if indicated in the security profile.

From Release 14 onwards, the default value of the enterprise parameter **OAuth Access Token for Devices** is **Implicit:Already registered devices**.

Step 3 Click **Save**.

Configure Refresh Logins

Use this procedure to configure Refresh Logins with OAuth access tokens and refresh tokens for Cisco Jabber clients.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Enterprise Parameters**.

Step 2 Under **SSO and OAuth Configuration**, set the **OAuth with Refresh Login Flow** parameter to **Enabled**.

Step 3 (Optional) Set any other parameters in the **SSO and OAuth Configuration** section. For parameter descriptions, click on the parameter name.

Step 4 Click **Save**.

Configure OAuth Ports

Use this procedure to assign the ports that are used for SIP OAuth.

Procedure

Step 1 From Cisco Unified CM Administration, choose, **System** > **Cisco Unified CM**.

Step 2 Do the following for each server that uses SIP OAuth.

Step 3 Select the server.

Step 4 Under Cisco Unified Communications Manager **TCP Port Settings**, set the port values for the following fields:

- SIP Phone OAuth Port

Default value is 5090. Acceptable configurable range is 1024–49151.

- SIP Mobile and Remote Access Port

  Default value is 5091. Acceptable configurable range is 1024–49151.

Note
Cisco Unified Communications Manager uses SIP Phone OAuth Port (5090) to listen for SIP line registration from Jabber on-premises devices over TLS. However, Unified CM uses SIP Mobile Remote Access Port (default 5091) to listen for SIP line registrations from Jabber over Expressway through mTLS.

Both ports use the Cisco Tomcat certificate and Tomcat-trust for incoming TLS/mTLS connections. Make sure that your Tomcat-trust store is able to verify the Expressway-C certificate for SIP OAuth mode for Mobile and Remote Access to function accurately.

You must perform extra steps to upload the Expressway-C certificate into the Tomcat-Trust certificate store of the Cisco Unified Communications Manager, when:

- Expressway-C certificate and Cisco Tomcat certificate is not signed by the same CA certificate.
- Unified CM Cisco Tomcat certificate is not CA signed.

Step 5 Click **Save**.

Step 6 Repeat this procedure for each server that uses SIP OAuth.

Configure OAuth Connection to Expressway-C

Use this procedure to add the Expressway-C connection to Cisco Unified Communications Manager Administration. You need this configuration for devices in Mobile and Remote Access mode with SIP OAuth.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Expressway-C**.

Step 2 (Optional) In the **Find and List Expressway-C** window, click **Find** to verify X.509 Subject Name/Subject Alternate Name that is pushed from the Expressway-C to Unified Communications Manager.

Note
If required, you can modify the values. Alternatively, if the entries are missing, add Expressway-C information.

If the Expressway-C has a different domain than the Unified Communications Manager, then the administrator needs to access the Cisco Unified CM Administration User Interface and add the domain to the Expressway C in the Unified CM configuration.

Step 3 Click **Add New**.

Step 4 Enter an IP Address, Hostname or fully qualified domain name for the Expressway-C.

Step 5 Enter a Description.

Step 6 Enter the X.509 Subject Name/Subject Alternate Name of the Expressway-C from the Expressway-C certificate.

Step 7 Click **Save**.

Enable SIP OAuth Mode

Use the Command Line Interface to enable SIP OAuth mode. Enabling this feature on the publisher node also enables the feature on all cluster nodes.

Before you begin

From Release 14SU1 onwards, when Proxy TFTP is enabled, you should copy the root CA certificate for the off-cluster Tomcat certificate to the proxy phone edge trust.

Procedure

Step 1 On the Unified Communications Manager publisher node, log in to the Command Line Interface.

Step 2 Run the `utils sipOAuth-mode enable` CLI command.
From Release 14 onwards, the system updates the read-only **Cluster SIPOAuth Mode** enterprise parameter to **Enabled**.

Restart Cisco CallManager Service

After enabling SIP OAuth through CLI, restart the Cisco CallManager service on all nodes where endpoints register through SIP OAuth.

Procedure

Step 1 From Cisco Unified Serviceability, choose **Tools** > **Control Center** > **Feature Services**.

Step 2 From the **Server** drop-down list, select the server.

Step 3 Check the **Cisco CallManager** service and click **Restart**.

Configure Device Security Mode in Phone Security Profile

Use this procedure to configure the device security mode in the phone security profile and is required only if you have set the **Device Security Mode** within that phone's **Phone Security Profile** to **Encrypted**.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Security** > **Phone Security Profile**.

Step 2 Perform either of the following:

- Search for an existing phone security profile
- Click **Add New**

Step 3 In the Phone Security Profile Information section, from the **Device Security Mode** drop-down list, choose **Encrypted**.

Step 4 From the **Transport Type** drop-down list, choose **TLS**.

Step 5 Check the **Enable OAuth Authentication** check box.

Step 6 Click **Save**.

Step 7 Associate the Phone Security Profile to the phone. For more information on how to apply the phone security phones, see "Apply Security Profiles to Phone" section in Security Guide for Cisco Unified Communications Manager.

Note
Reset your phone for the changes to take effect.

Note
When SIP OAuth Mode is enabled, **Enable Digest Authentication** and **TFTP Encrypted Config** options are not supported. Phones will download the TFTP config file securely over **https(6971)** and use the token for authentication.

Configure SIP Oauth Registered Phones for MRA Mode

Use this procedure to configure SIP OAuth registered phones to MRA mode.

Before you begin

Important This section is applicable from Release 14 onwards.

Make sure your phones are configured to use Activation Codes. For more information see *Set Registration Method to use Activation Codes* section in System Configuration Guide for Cisco Unified Communications Manager.

Note When using SIP OAuth over MRA , user cannot use username / password for login but have to use activation code based onboarding

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.

Step 2 Click **Find** and select the device which you want to configure for off-premises mode.

Step 3 In the **Device Information** section, do the following:

- Check **Allow Activation Code via MRA** check box.
- From the **Activation Code MRA Service Domain** drop-down list, choose the required MRA service domain. For more information on how to configure the MRA service domain see, the *MRA Service Domain Configuration* section in System Configuration Guide for Cisco Unified Communications Manager.

Note
For SIP OAuth over MRA mode, use only activation code and do not use username/password based login.

Step 4 In the **Protocol Specific Information** section, choose the OAuth enabled SIP profile from the **Device Security Profile** drop-down list. Make sure that the phone supports OAuth firmware. For more information, on how to create a security profile, see *Configure Phone Security Profile* section in System Configuration Guide for Cisco Unified Communications Manager.

Step 5 Click **Save** and **Apply Configuration**.

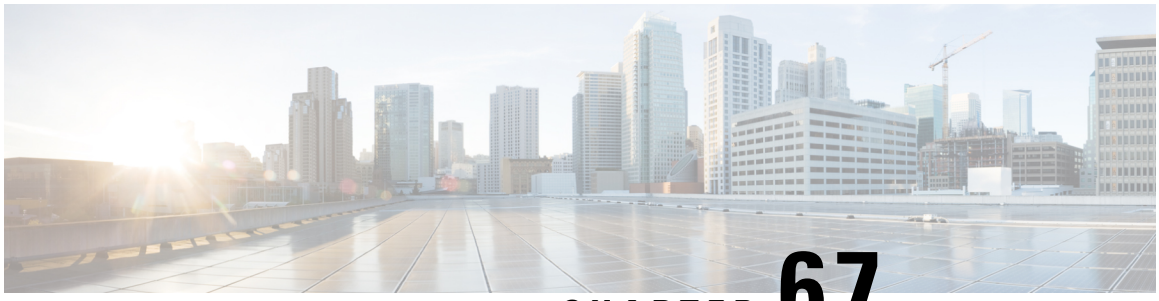
Note
The phone switches to MRA mode and initiates communication with the Expressway. If your internal network does not allow communication with Expressway from on-premises, the phone doesn't register but is ready to contact Expressway when it's powered up off-premises.



PART XV

QoS Management

- Configure AS-SIP Endpoints, on page 813
- Configure Multilevel Precedence and Preemption, on page 825



CHAPTER 67

Configure AS-SIP Endpoints

- AS-SIP Overview, on page 813
- AS-SIP Prerequisites, on page 815
- AS-SIP Enpdoint Configuration Task Flow, on page 816

AS-SIP Overview

Assured Services SIP (AS-SIP) endpoints are compliant with MLPP, DSCP, TLS/SRTP, and IPv6 requirements. AS-SIP provides for multiple endpoint interfaces on the Unified Communications Manager.

Many Cisco IP phones support AS-SIP. In addition, the Third-Party AS-SIP Endpoint device type allows a third-party AS-SIP compliant endpoint to be configured and used with Cisco Unified Communications Manager. In addition, the Third-Party AS-SIP Endpoint device type allows a third-party AS-SIP-compliant generic endpoint to be configured and used with Cisco Unified Communications Manager

AS-SIP Capabilities

The following capabilities are implemented or made available for AS-SIP endpoints:

- MLPP
- TLS
- SRTP
- DSCP for precedence levels
- Error responses
- V.150.1 MER
- Conference Factory flow support
- AS-SIP Line Early Offer

Third-Party AS-SIP Phones

Third-party phones can be provisioned in Cisoc Unified Communications Manager using the Third-Party AS-SIP Endpoint device type.

Third-party phones that are running AS-SIP do not get configured through the Cisco Unified Communications Manager TFTP server. The customer must configure them by using the native phone configuration mechanism (usually a web page or TFTP file). The customer must keep the device and line configuration in the Cisco Unified Communications Manager database synchronized with the native phone configuration (for example, extension 1002 on the phone and 1002 in Cisco Unified Communications Manager). Also, if the directory number of a line is changed, the customer must ensure that it gets changed in both Cisco Unified CM Administration and in the native phone configuration mechanism.

Identification of Third-Party Phones

The third-party phones that are running SIP do not send a MAC address, they must identify themselves by using username. The REGISTER message includes the following header:

```
Authorization: Digest
username="swhite",realm="ccmsipline",nonce="GBauADss2qoWr6k9y3hGGVDAqnLfoLk5",uri
="sip:172.18.197.224",
algorithm=MD5,response="126c0643a4923359ab59d4f53494552e"
```

The username, **swhite**, must match a user that is configured in the **End User Configuration** window of Cisco Unified CM Administration. The administrator configures the SIP third-party phone with the user; for example, **swhite**, in the **Digest User** field of **Phone Configuration** window.

Note You can assign each user ID to only one third-party phone. If the same user ID is assigned as the Digest User for multiple phones, the third-party phones to which they are assigned will not successfully register.

Configuration of Third Party AS-SIP Phones and Cisco IP Phones

The following table provides a comparison overview of the configuration differences between Cisco Unified IP Phones and third-party phones that are running AS-SIP.

Table 83: Comparison of the Configuration Differences Between Cisco IP Phones and Third-Party Phones

Phone Running AS-SIP	Integrated with Centralized TFTP	Sends MAC Address	Downloads Softkey File	Downloads Dial Plan File	Supports Unified Communications Manager Failover and Fallback	Supports Reset and Restart
Cisco IP Phone	Yes	Yes	Yes	Yes	Yes	Yes
Third-party AS-SIP device	No	No	No	No	No	No

Note Not all Cisco IP Phones support AS-SIP. See the phone administration guide for your phone model for support information

Use Cisco Unified CM Administration to configure third-party phones that are running SIP (For more information, see "Configure SIP Profile" topic in *System Configuration Guide for Cisco Unified Communications Manager*

the). The administrator must perform configuration steps on the third-party phone that is running SIP; see the following examples:

- Ensure that proxy address in the phone is the IP or Fully Qualified Domain Name (FQDN) of Cisco Unified Communications Manager.
- Ensure directory numbers in the phone match the directory numbers that are configured for the device in Cisco Unified CM Administration.
- Ensure digest user ID (sometimes referred to as Authorization ID) in the phone matches the Digest User ID in the Cisco Unified CM Administration.

For more information, refer to the documentation that came with the third-party phone.

AS-SIP Conferencing

MOH is applied to its target (a held party, transferee just before transfer, or conferee just before joining the conference), if the feature invoker (holder, transferor, or conference initiator) supports Cisco-proprietary feature signaling. If the feature invoker does not support Cisco-proprietary feature signaling, then MOH is not applied to its target. Also, if an endpoint explicitly signals that it is a conference mixer, then MOH will not be played to the target. There are two forms of AS-SIP Conferencing:

- Local mixing
- Conference Factory

Local mixing

To the Unified CM, the conference initiator simply appears to have established simultaneously active calls, one to each of the other conference attendees. The initiator host the conference locally and the voices are mixed there. The calls from the conference initiator have special signaling that prevent it from being connected to an MOH source.

Conference Factory

The conference initiator calls a Conference Factory Server located off a SIP trunk. Through IVR signaling, the conference initiator instructs the Conference Factory to reserve a conference bridge. The Conference Factory gives the numeric address (a routable DN) to the conference initiator, who then establishes a subscription with the bridge to receive conference list information to track the participants. The Conference Factory sends special signaling that prevent it from being connected to an MOH Source.

AS-SIP Prerequisites

Determine whether sufficient Device License Units are available. For more information, see"Smart Software Licensing" chapter from *System Configuration Guide for Cisco Unified Communications Manager*

AS-SIP Enpdoint Configuration Task Flow

Complete the following tasks to configure an AS-SIP endpoint.

Procedure

	Command or Action	Purpose
Step 1	Configure a Digest User, on page 817	Configure the end user to use digest authentication for SIP requests.
Step 2	Configure SIP Phone Secure Port, on page 817	Cisco Unified Communications Manager uses this port to listen to SIP phones for SIP line registrations over TLS.
Step 3	Restart Services, on page 817	After configuring the secure port, restart the Cisco CallManager and Cisco CTL Provider services.
Step 4	Configure SIP Profile for AS-SIP, on page 818	Configure a SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks. **Note** The phone-specific parameters are not downloaded to a third-party AS-SIP phone. They are used only by Cisco Unified Communications Manager. Third-party phones must locally configure the same settings.
Step 5	Configure Phone Security Profile for AS-SIP, on page 819	You can use the phone security profile to assign security settings such as TLS, SRTP, and digest authentication
Step 6	Configure AS-SIP Endpoint, on page 819	Configure a Cisco IP Phone or a third-party endpoint with AS-SIP support.
Step 7	Associate Device with End User, on page 820	Associate the endpoint with a user.
Step 8	Configure SIP Trunk Security Profile for AS-SIP, on page 821	You can use the sip trunk security profile to assign security features such as TLS or digest authentication to a SIP trunk.
Step 9	Configure SIP Trunk for AS-SIP, on page 821	Configure a SIP trunk with AS-SIP support.
Step 10	Configure AS-SIP Features, on page 822	Configure additional AS-SIP features such as MLPP, TLS, V.150 and IPv6.

Configure a Digest User

Use this procedure to configure an end user as a digest user whom uses digest authentication. Devices that are associated to the user will be authenticated via the user's digest credentials.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **End User**.

Step 2 Do either of the following:

- Click **Add New** to create a new user.
- Click **Find** and select an existing user.

Step 3 Make sure the following mandatory fields are completed:

- User ID
- Last Name

Step 4 In the **Digest Credentials** field, enter a password. End users must authenticate themselves via this password when using the endpoint.

Step 5 Complete any remaining fields. For help with the fields and their settings, see the online help.

Step 6 Click **Save**.

Configure SIP Phone Secure Port

Follow these steps to configure the SIP Phone Secure Port. Cisco Unified Communications Manager uses this port to listen to SIP phones for SIP line registrations over TLS.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Cisco Unified CM**.

Step 2 In the **Cisco Unified Communications Manager TCP Port Settings for this Server** section, specify a port number in the **SIP Phone Secure Port** field, or leave the field set to default. The default value is 5061.

Step 3 Click **Save**.

Step 4 Click **Apply Config**.

Step 5 Click **Ok**.

Restart Services

Follow these steps to restart Cisco CallManager and Cisco CTL Provider services.

Procedure

Step 1 From the Cisco Unified Serviceability interface, choose **Tools** > **Control Center - Feature Services**.

Step 2 Choose the Cisco Unified Communications Manager server from the **Servers** drop-down list.
In the CM Services area, Cisco CallManager displays in the **Service Name** column.

Step 3 Click the radio button that corresponds to the Cisco CallManager service.

Step 4 Click **Restart**.
The service restarts and displays the message, `Service Successfully Restarted`.

Step 5 Repeat step 3 and step 4 to restart Cisco CTL Provider service.

Configure SIP Profile for AS-SIP

Use this procedure to configure SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks.

Procedure

Step 1 In Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.

Step 2 Do either of the following:

- Click **Add New** to create a new SIP Profile.
- Click **Find** and select an existing SIP Profile.

Step 3 Enter a **Name** and **Description** for the SIP Profile.

Step 4 Check the **Assured Services SIP conformance** check box.

Note
This checkbox must be checked for SIP trunks and for third-party AS-SIP phones. It's not mandatory for Cisco IP Phones that support AS-SIP.

Step 5 In the **Parameters used in Phone** section, configure DSCP precedence values for the types of calls that you expect to make.

Note
You can also configure DSCP values via clusterwide service parameters. However, the DSCP values within a SIP Profile override the clusterwide settings for all devices that use the SIP Profile.

Step 6 From the **Early Offer support for voice and video calls** drop-down list, select one of the following options to configure Early Offer support for SIP trunks that use this profile:

- Disabled
- Best Effort (no MTP Inserted)
- Mandatory (insert MTP if needed)

Step 7 Complete the remaining fields in the **SIP Profile Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 8 Click **Save**.

Configure Phone Security Profile for AS-SIP

Use this procedure to configure a phone security profile for AS-SIP endpoints. You can use the security profile to assign security settings such as TLS and SRTP.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Security** > **Phone Security Profile**.

Step 2 Perform one of the following steps:

- Click **Add New** to create a new phone security profile.
- Click **Find** to edit an existing profile.

Step 3 For new profiles, select an option from the **Phone Security Profile** drop-down, choose the phon emodel**Third-party AS-SIP Endpoint** and click **Next**.

- For Cisco IP phones, select the phone model and click **Next**.
- For third-party AS-SIP endpoints, select **Third-party AS-SIP Endpoint** and click **Next**.

Step 4 For the protocol, select **SIP** and click **Next**.

Step 5 Enter a **Name** and **Description** for the protocol.

Step 6 Assign the **Device Security Mode**, to one of the following settings:

- **Authenticated**—Cisco Unified Communications Manager uses TLS signaling, providing integrity and authentication for the phone.
- **Encrypted**—Cisco Unified Communications Manager uses TLS signaling, providing integrity and authentication for the phone. In addition, SRTP encrypts the media streams.

Step 7 Check the **Enable Digest Authentication** check box.

Step 8 Configure the remaining fields in the **Phone Security Profile Configuration** window. For help with the fields and their settings, see the online help.

Step 9 Click **Save**.

Configure AS-SIP Endpoint

Use this procedure to configure an AS-SIP endpoint. Many Cisco IP Phones support AS-SIP. In addition, you can configure AS-SIP for third-party endpoints.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.

Step 2 Click **Add New**.

Step 3 From the Phone Type drop-down list, select a Cisco IP Phone that supports AS-SIP. Otherwise, select **Third-Party AS-SIP Endpoint**.

Step 4 Click **Next**.

Step 5 Configure the following mandatory fields.For more information on the fields and their configuration options, see Online Help.

- Device Trust Mode—For third-party AS-SIP endpoints only. Select **Trusted** or **Not Trusted**.
- MAC Address
- Device Pool
- Phone Button Template
- Owner User ID
- Device Security Profile—Select the phone security profile that you set up for AS-SIP.
- SIP Profile—Select the AS-SIP-enabled SIP Profile that you configured.
- Digest User—Select the user ID that you configure as a digest user. The user must be enabled for digest authentication
- Require DTMF Reception—Check this check box to allow the endpoint to accept DTMF digits.
- Early Offer support for voice and video calls—Check this check box to enable early offer support. This field appears for third-party phones only.

Step 6 Configure the fields in the **MLPP and Confidential Access Level Information** section.

Step 7 Click **Save**.

Step 8 Add a Directory Number:

a) In the left navigation bar, click **Add a new DN**. The **Directory Number Configuration** window opens.
b) Add a **Directory Number**.
c) Complete any remaining fields in the **Directory Number Configuration** window
d) Click **Save**.

Step 9 From **Related Links**, select **Configure Device** and click **Go**.

Step 10 Click **Apply Config**.

Associate Device with End User

Use this procedure to associate an end user to the AS-SIP endpoint.

Procedure

Step 1 From Cisco Unified CM Administration, choose **User Management** > **End User**.

Step 2 Click **Find** and select the user whom you want to associate to the device.

Step 3 In the **Device Information** section, click **Device Association**.

The User Device Association window appears.

Step 4 Click **Find** to view a list of available devices.

Step 5 Select the device that you want to associate, and click **Save Selected/Changes**.

Step 6 From **Related Links**, choose **Back to User**, and click **Go**.
The **End User Configuration** window appears, and the associated device that you chose appears in the **Controlled Devices** pane.

Configure SIP Trunk Security Profile for AS-SIP

Use this procedure to configure a security profile for a SIP trunk that supports AS-SIP

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **Security** > **SIP Trunk Security Profile**.

Step 2 Click **Add New**.

Step 3 Enter a **Name** for the security profile.

Step 4 From the **Device Security Mode** drop-down list, choose **Authenticated** or **Encrypted**.

Step 5 The **Incoming Transport Type** and **Outgoing Transport Type** fields change to **TLS** automatically.

Step 6 Check the **Enable Digest Authentication** check box.

Step 7 If you are deploying V.150, configure a value for the **SIP V.150 Outbound SDP Offer Filtering** drop-down list.

Step 8 Complete the remaining fields in the **SIP Trunk Security Profile Configuration** window.For more information on the fields and their configuration options, see Online Help.

Step 9 Click **Save**.

Configure SIP Trunk for AS-SIP

Use this procedure to set up a SIP trunk that supports AS-SIP.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Trunk**.

Step 2 Do either of the following:

- Click **Find** and select an existing trunk.
- Click **Add New** to create a new trunk.

Step 3 For new trunks, from the **Trunk Type** drop-down list, select **SIP Trunk**.

Step 4 From the **Trunk Service Type** drop-down list, select **None (Default)** and click **Next**.

Step 5 Enter a **Device Name** for the trunk.

Step 6 From the **Device Pool** drop-down list, select a device pool.

Step 7 In the **Destination Address** field, enter the address of the server to which you are connecting the trunk.

Step 8 From the **SIP Trunk Security Profile** drop-down list, select the profile that you created for AS-SIP.

Step 9 From the **SIP Profile** drop-down list, select the SIP Profile that you set up for AS-SIP.

Step 10 Complete any remaining fields in the **Trunk Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 11 Click **Save**.

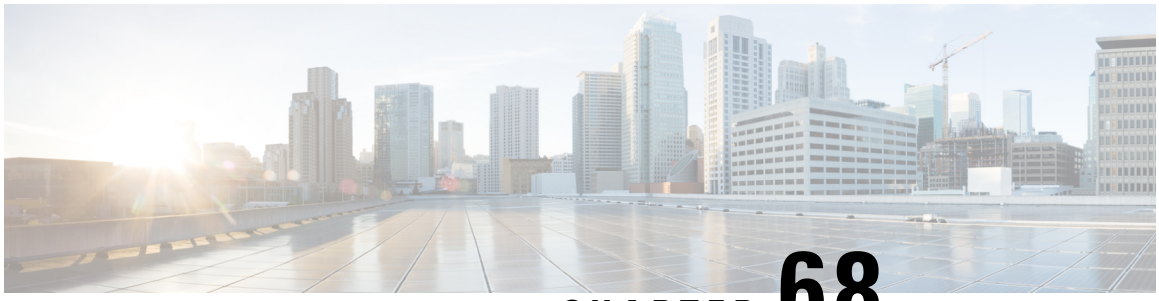
Configure AS-SIP Features

The procedures in the preceding task flow describe how to configure AS-SIP support on endpoints and trunk. The following table outlines the AS-SIP features that you can deploy and provides configuration reference for each.

AS-SIP Feature	Configuration Description
Early Offer	SIP Early Offer allows your endpoints to negotiate media during the INVITE request and the 200OK response. There are two modes for Early Offer: • Best Effort Early Offer (no MTP Inserted) • Mandatory Early Offer (insert MTP if needed) Configure Early Offer support via the fields in the following configuration windows. Refer to the online help for detailed field descriptions: **SIP Profile Configuration** window • Early Offer support for voice and video calls—Configure this field to enable Early Offer support on a SIP trunk • SDP Session-level Bandwidth Modifier for Early Offer and Re-invite • Send send-receive SDP in mid-call INVITE **Phone Configuration** window (only if the Third Pary AS-SIP Endpoint device type is used) • Early Offer support for voice and video calls - check this check box to enable early offer support

AS-SIP Feature	Configuration Description
Conference Factory	Specify the URI that an IMS client uses to set up a conference: 1. From Cisco Unified CM Administration, choose **System** > **Service Parameters**. 2. From the **Server** drop-down list select your Cisco Unified Communications Manager server. 3. From the **Service**, select **Cisco CallManager**. 4. Under **Clusterwide Paramters (Feature - Conference)** assign an **IMS Conference Factory URI**. 5. Click **Save**.
DSCP Markings	DSCP settings allow you to manage QoS and bandwidth within your network. DSCP settings are used to assign a prioritized Traffic Class Label to calls on a per-call basis. You can configure clusterwide DSCP settings via service parameters and you can use the SIP Profile to assign a customized QoS policy for users whom use that profile. For example, you could assign higher priority for the calls of an executive (for example, a CEO) or a sales team to ensure that their calls are not dropped if network bandwidth issues arise. To configure DSCP, see DSCP Settings Configuration Task Flow, on page 779.
IPv6	By default, Cisco Unified Communications Manager is configured to use IPv4 addressing. However, you can configure the system to support the IPv6 stack thereby allowing you to deploy a SIP network with IPv6-only endpoints. For more information to configure IPv6, see "Dual Stack IPv6 Configuration Task Flow" chapter in the *System Configuration Guide for Cisco Unified Communications Manager*
Multilevel Precedence and Preemption (MLPP)	The Multilevel Precedence and Preemption (MLPP) service allows placement of priority calls. This capability assures high-ranking personnel of communication to critical organizations and personnel during network stress situations, such as a national emergency or degraded network situations. To configure MLPP, see Multilevel Precendence and Preemption Task Flow, on page 825.
Secure Real-Time Transport Protocol (SRTP)	The Secure Real-time Transport Protocol (SRTP) can be used to provide encryption and authentication to media streams in your calls. SRTP can be configured for phones within the **Phone Security Profile Configuration** that the phone uses. You must set the **Device Security Mode** field to **Encrypted**.

AS-SIP Feature	Configuration Description
Transport Layer Signalling (TLS)	Transport Layer Security (TLS) provides secure and reliable signaling and data transfer between two systems or devices, by using secure ports and certificate exchange. For more information to configure TLS, see the "TLS Setup" chapter in the *Security Guide for Cisco Unified Communications Manager*.
V.150	The V.150 Minimum Essential Requirements feature allows you to make secure calls in a modem over IP network. The feature uses a dialup modem for large installed bases of modems and telephony devices operating on a traditional public switched telephone network (PSTN). For more information to configure V.150, see the "Cisco V.150 Minimum Essential Requirements (MER)" chapter in the *Security Guide for Cisco Unified Communications Manager*.



CHAPTER 68

Configure Multilevel Precedence and Preemption

- Multilevel Precedence and Preemption Overview, on page 825
- Multilevel Precedence and Preemption Prerequisites, on page 825
- Multilevel Precendence and Preemption Task Flow, on page 825
- Multilevel Precedence and Preemption Interactions, on page 841
- Multilevel Precedence and Preemption Restrictions, on page 842

Multilevel Precedence and Preemption Overview

The Multilevel Precedence and Preemption (MLPP) service allows placement of priority calls. Properly validated users can preempt lower priority phone calls with higher priority calls. An authenticated user can preempt calls either to targeted stations or through fully subscribed TDM trunks. This capability assures high-ranking personnel of communication to critical organizations and personnel during network stress situations, such as a national emergency or degraded network situations.

Multilevel Precedence and Preemption Prerequisites

Supported SCCP or SIP phones. See the *Cisco IP Phone Administration Guide* and *Cisco IP Phone User Guide* for your phones for feature support and more information.

Multilevel Precendence and Preemption Task Flow

Before you begin

Procedure

	Command or Action	Purpose
Step 1	To Configure Domains and Domain Lists, on page 827, perform the following subtasks: • Configure a Multilevel Precedence and Preemption Domain, on page 828	Configure an MLPP domain to specify the devices and resources that are associated with an MLPP subscriber.

	Command or Action	Purpose
	• Configure a Resource Priority Namespace Network Domain, on page 828 • Configure a Resource Priority Namespace Network Domain List, on page 829	
Step 2	Configure a Common Device Configuration for Multilevel Precedence and Preemption, on page 829	A common device configuration includes MLPP-related information that can be applied to multiple users and their devices. Ensure that each device is associated with a common device configuration. These settings override the enterprise parameter settings.
Step 3	Configure the Enterprise Parameters for Multilevel Precedence and Preemption, on page 830	Set enterprise parameters to enable MLPP indication and preemption. If individual devices and devices in common device configurations have MLPP settings of Default, the MLLP-related enterprise parameters apply to these devices and common device configurations.
Step 4	Configure a Partition for Multilevel Precedence and Preemption, on page 831	Configure a partition to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Devices that are typically placed in partitions include DNs and route patterns. These entities associate with DNs that users dial. For simplicity, partition names usually reflect their characteristics.
Step 5	Configure a Calling Search Space for Multilevel Precedence and Preemption, on page 832	A calling search space is an ordered list of partitions. Calling search spaces determine the partitions that calling devices, including IP phones, softphones, and gateways, can search when attempting to complete a call.
Step 6	Configure a Route Pattern for Multilevel Precedence and Preemption, on page 833	Configure route patterns to route or block both internal and external calls.
Step 7	Configure a Translation Pattern for Multilevel Precedence and Preemption, on page 834	Configure translation patterns to specify how to route a call after it is placed. Configuring translation patterns allows your system to manipulate calling and called digits as needed. When the system identifies that a pattern match occurred, your system uses the calling search space that is configured for the translation pattern to perform the subsequent match.
Step 8	Configure Multilevel Precedence and Preemption for Gateways, on page 835	Configure Cisco Unified Communications Manager to communicate with non-IP telecommunications devices.

	Command or Action	Purpose
Step 9	Configure Multilevel Precedence and Preemption for Phones, on page 836	
Step 10	Configure a Directory Number to Place Multilevel Precedence and Preemption Calls, on page 838	After you configure a device, you can add a line (directory number) from the updated **Device Configuration** window.
Step 11	Configure a User Device Profile for Multilevel Precedence and Preemption, on page 839	When a user profile is assigned to a phone, the phone inherits the configuration of the assigned user, including any CSS that is associated with the user. The phone CSS can, however, override the user profile. Cisco Unified Communications Manager assigns the precedence level that is associated with the dialed pattern to the call when a pattern match occurs. The system sets the call request as a precedence call with the assigned precedence level.
Step 12	Configure the Default Device Profile for Multilevel Precedence and Preemption, on page 840	Use the default device profile for whenever a user logs on to a phone model for which no user device profile exists. A default device profile comprises the set of services and features that are associated with a particular device.

Configure Domains and Domain Lists

Configure an MLPP domain to specify the devices and resources that are associated with an MLPP subscriber.

Procedure

	Command or Action	Purpose
Step 1	Configure a Multilevel Precedence and Preemption Domain, on page 828	Associate devices and resources with an MLPP subscriber. When an MLPP subscriber that belongs to a particular domain places a precedence call to another MLPP subscriber that belongs to the same domain, the MLPP service can preempt the existing call that the called MLPP subscriber is on for a higher precedence call. MLPP service availability does not span across different domains. The MLPP domain subscription of the originating user determines the domain of the call and its connections. Only higher precedence calls in one domain can preempt connections that calls in the same domain are using.

	Command or Action	Purpose
Step 2	Configure a Resource Priority Namespace Network Domain, on page 828	Configure namespace domains for a Voice over Secured IP (VoSIP) network that uses SIP trunks. Your system prioritizes the SIP-signaled resources so that those resources can be used most effectively during emergencies and congestion of telephone circuits, IP bandwidth, and gateways. Endpoints receive the precedence and preemption information.
Step 3	Configure a Resource Priority Namespace Network Domain List, on page 829	Configure a list of acceptable network domains. Incoming calls are compared to the list and processed, if an acceptable network domain is in the list.

Configure a Multilevel Precedence and Preemption Domain

Associate devices and resources with an MLPP subscriber. When an MLPP subscriber that belongs to a particular domain places a precedence call to another MLPP subscriber that belongs to the same domain, the MLPP service can preempt the existing call that the called MLPP subscriber is on for a higher precedence call. MLPP service availability does not span across different domains.

The MLPP domain subscription of the originating user determines the domain of the call and its connections. Only higher precedence calls in one domain can preempt connections that calls in the same domain are using.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **MLPP** > **Domain** > **MLPP Domain**.

Step 2 Click **Add New**.

Step 3 In the **Domain Name** field, enter the name that you want to assign to the new MLPP domain.

You can enter up to 50 alphanumeric characters, and any combination of spaces, periods (.), hyphens (-), and underscore characters (_).

Step 4 In the **Domain ID** field, enter a unique six-character hexadecimal MLPP domain ID.

Domain IDs must fall in the range between 000001 and FFFFFF. (000000 is reserved for the default MLPP domain ID.)

Step 5 Click **Save**.

Configure a Resource Priority Namespace Network Domain

Configure namespace domains for a Voice over Secured IP (VoSIP) network that uses SIP trunks. Your system prioritizes the SIP-signaled resources so that those resources can be used most effectively during emergencies and congestion of telephone circuits, IP bandwidth, and gateways. Endpoints receive the precedence and preemption information.

Procedure

Step 1 From Cisco Unified CM Administration, **System** > **MLPP** > **Namespace** > **Resource Priority Namespace Network Domain**.

Step 2 Enter the name for the Resource Priority Namespace Network Domain in the information section. The maximum number of domain names is 100.

Step 3 Enter a description for the domain name.

The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), or angle brackets (<>).

Step 4 Check the **Make this the Default Resource Priority Namespace Network Domain** check box if you want the domain name to be the default.

Step 5 Click **Save**.

Configure a Resource Priority Namespace Network Domain List

Configure a list of acceptable network domains. Incoming calls are compared to the list and processed, if an acceptable network domain is in the list.

Procedure

Step 1 From Cisco Unified CM Administration, choose **System** > **MLPP** > **Namespace** > **Resource Priority Namespace List**.

Step 2 Enter the name for the Resource Priority Namespace List. The maximum number of characters is 50.

Step 3 Enter a description for the list. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\), or angle brackets (<>).

Step 4 Use the Up and Down Arrows to move a Resource Priority Namespace Network Domain to the **Selected Resource Priority Namespaces** field.

Step 5 Click **Save**.

Configure a Common Device Configuration for Multilevel Precedence and Preemption

A common device configuration includes MLPP-related information that can be applied to multiple users and their devices. Ensure that each device is associated with a common device configuration. These settings override the enterprise parameter settings.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Common Device Configuration**.

Step 2 Perform one of the following tasks:

- Click **Find** to modify an existing common device configuration and choose a common device configuration from the resulting list.
- Click **Add New** to add a new common device configuration.

Step 3 Configure the fields on the **Common Device Configuration** window. For more information on the fields and their configuration options, see Online Help.

Step 4 Click **Save**.

Configure the Enterprise Parameters for Multilevel Precedence and Preemption

Set enterprise parameters to enable MLPP indication and preemption. If individual devices and devices in common device configurations have MLPP settings of Default, the MLLP-related enterprise parameters apply to these devices and common device configurations.

Procedure

Step 1 Choose **System** > **Enterprise Parameters**.

Step 2 Configure the MLPP enterprise parameters on the **Enterprise Parameters Configuration** window. See the Related Topics section for more information about the parameters and their configuration options.

Step 3 Click **Save**.

Enterprise Parameters for Multilevel Precedence and Preemption

Table 84: Enterprise Parameters for Multilevel Precedence and Preemption

Parameter	Description
MLPP Domain Identifier	Set this parameter to define a domain. Because MLPP service applies to a domain, Cisco Unified Communications Manager marks only connections and resources that belong to calls from MLPP users in a given domain with a precedence level. Cisco Unified Communications Manager can preempt only lower precedence calls from MLPP users in the same domain. The default is **000000**.

Parameter	Description
MLPP Indication Status	This parameter specifies whether devices use MLPP tones and special displays to indicate MLPP precedence calls. To enable MLPP indication across the enterprise, set this parameter to MLPP Indication turned on. The default is **MLPP Indication turned off**.
MLPP Preemption Setting	This parameter determines whether devices should apply preemption and preemption signaling (such as preemption tones) to accommodate higher precedence calls. To enable MLPP preemption across the enterprise, set this parameter to Forceful Preemption. The default is **No preemption allowed**.
Precedence Alternate Party Timeout	In a precedence call, if the called party subscribes to alternate party diversion, this timer indicates the seconds after which Cisco Unified Communications Manager will divert the call to the alternate party if the called party does not acknowledge preemption or does not answer a precedence call. The default is **30** seconds.
Use Standard VM Handling For Precedence Calls	This parameter determines whether a precedence call will forward to the voice-messaging system. If the parameter is set to False, precedence calls do not forward to the voice-messaging system. If the parameter is set to True, precedence calls forward to the voice-messaging system. For MLPP, the recommended setting for this parameter is False, as users, not the voice-messaging system, should always answer precedence calls. The default is **False**.

Configure a Partition for Multilevel Precedence and Preemption

Configure a partition to create a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Devices that are typically placed in partitions include DNs and route patterns. These entities associate with DNs that users dial. For simplicity, partition names usually reflect their characteristics.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Class of Control** > **Partition**.

Step 2 Click **Add New** to create a new partition.

Step 3 In the **Partition Name, Description** field, enter a name for the partition that is unique to the route plan.

Partition names can contain alphanumeric characters, as well as spaces, hyphens (-), and underscore characters (_). See the online help for guidelines about partition names.

Step 4 Enter a comma (,) after the partition name and enter a description of the partition on the same line.

The description can contain up to 50 characters in any language, but it cannot include double quotes ("), percentage sign (%), ampersand (&), backslash (\), angle brackets (<>), or square brackets ([ ]).

If you do not enter a description, Cisco Unified Communications Manager automatically enters the partition name in this field.

Step 5 To create multiple partitions, use one line for each partition entry.

Step 6 From the **Time Schedule** drop-down list, choose a time schedule to associate with this partition.

The time schedule specifies when the partition is available to receive incoming calls. If you choose **None**, the partition remains active at all times.

Step 7 Select one of the following radio buttons to configure the **Time Zone**:

- **Originating Device**—When you select this radio button, the system compares the time zone of the calling device to the **Time Schedule** to determine whether the partition is available 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**.

Partition Naming Guidelines

The list of partitions in a calling search space is limited to a maximum of 1024 characters. This means that the maximum number of partitions in a CSS varies depending on the length of the partition names. Use the following table to determine the maximum number of partitions that you can add to a calling search space if partition names are of fixed length.

Table 85: Partition Name Guidelines

Partition Name Length	Maximum Number of Partitions
2 characters	340
3 characters	256
4 characters	204
5 characters	172
...	...
10 characters	92
15 characters	64

Configure a Calling Search Space for Multilevel Precedence and Preemption

A calling search space is an ordered list of partitions. Calling search spaces determine the partitions that calling devices, including IP phones, softphones, and gateways, can search when attempting to complete a call.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Class of Control** > **Calling Search Space**.

Step 2 Click **Add New**.

Step 3 In the **Name** field, enter a name.

Ensure that each calling search space name is unique to the system. The name can include up to 50 alphanumeric characters and can contain any combination of spaces, periods (.), hyphens (-), and underscore characters (_).

Step 4 In the **Description** field, enter a description.

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 a Route Pattern for Multilevel Precedence and Preemption

Configure route patterns to route or block both internal and external calls.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Call Routing** > **Route/Hunt** > **Route Pattern**.

Step 2 Perform one of the following tasks:

- To modify the settings for an existing route pattern, enter search criteria, click **Find**, and then choose an existing route pattern from the resulting list.
- To add a new route pattern, click **Add New**.

Step 3 Configure the fields on the **Route Pattern Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

Step 4 Click **Save**.

Route Pattern Configuration Fields for Multilevel Precedence and Preemption

Table 86: Route Pattern Configuration Fields for Multilevel Precedence and Preemption

Field	Description
Route Pattern	Enter the route pattern, including numbers and wildcards, without spaces. For example, for NANP, enter 9.@ for typical local access or 8XXX for a typical private network numbering plan. Valid characters include the uppercase characters A, B, C, and D and \+, which represents the international escape character +.
MLPP Precedence	Choose an MLPP precedence setting for this route pattern from the drop-down list: • Executive Override—Highest precedence setting for MLPP calls. • Flash Override—Second highest precedence setting for MLPP calls. • Flash—Third highest precedence setting for MLPP calls. • Immediate—Fourth highest precedence setting for MLPP calls. • Priority—Fifth highest precedence setting for MLPP calls. • Routine—Lowest precedence setting for MLPP calls. • Default-—Does not override the incoming precedence level but rather lets it pass unchanged.
Apply Call Blocking Percentage	Check this check box to enable the Destination Code Control (DCC) feature. By enabling DCC, all calls other than flash and higher precedence calls made to the destination are filtered and allowed or disallowed based on the Call Blocking Percentage quota set for the destination. Flash and higher precedence calls are allowed at all times. DCC is disabled by default. The Apply Call Blocking Percentage field is enabled only if the MLPP level is immediate, priority, routine or default.
Call Blocking Percentage (%)	Enter the percentage of calls to be blocked for this destination in numerals. This value specifies the percentage of lower precedence calls made to this destination that get blocked by the route pattern. This percentage limits the lower precedence calls only; the flash and higher precedence calls made to this destination are allowed at all times The Call Blocking Percentage (%) field is enabled only if the Apply Call Blocking Percentage check box is checked.
Resource Priority Namespace Network Domain	Choose a Resource Priority Namespace Network Domain from the drop-down list. To configure the Resource Priority Namespace Network Domains, choose System > MLPP > Namespace > Resource Priority Namespace Network Domain.

Configure a Translation Pattern for Multilevel Precedence and Preemption

Configure translation patterns to specify how to route a call after it is placed. Configuring translation patterns allows your system to manipulate calling and called digits as needed. When the system identifies that a pattern

match occurred, your system uses the calling search space that is configured for the translation pattern to perform the subsequent match.

Procedure

Step 1 In Cisco Unified CM Administration, choose **Call Routing** > **Translation Pattern**.

Step 2 Perform one of the following tasks:

- To modify the settings for an existing translation pattern, enter search criteria, click **Find**, and choose an existing Translation Pattern from the resulting list.
- To add a new translation pattern, click **Add New**.

Step 3 From the **MLPP Precedence** drop-down list, choose one of the following settings for this translation pattern:

- **Executive Override**—Highest precedence setting for MLPP calls.
- **Flash Override**—Second highest precedence setting for MLPP calls.
- **Flash**—Third highest precedence setting for MLPP calls.
- **Immediate**—Fourth highest precedence setting for MLPP calls.
- **Priority**—Fifth highest precedence setting for MLPP calls.
- **Routine**—Lowest precedence setting for MLPP calls.
- **Default**—Does not override the incoming precedence level but rather lets it pass unchanged.

Step 4 From the **Resource-Priority Namespace Network Domain** drop-down list, choose a resource priority namespace network domain that you configured.

Step 5 From the **Calling Search Space** drop-down list, choose the calling search space that you configured.

Step 6 Click **Save**.

Configure Multilevel Precedence and Preemption for Gateways

Configure Cisco Unified Communications Manager to communicate with non-IP telecommunications devices.

Before you begin

- Configure one of the following gateways:
  - Cisco Catalyst 6000 24 port FXS Gateway
  - Cisco Catalyst 6000 E1 VoIP Gateway
  - Cisco Catalyst 6000 T1 VoIP Gateway
  - Cisco DE-30+ Gateway
  - Cisco DT-24+ Gateway
  - H.323 Gateway

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Gateway**

Step 2 Perform one of the following tasks:

- To modify the settings for an existing gateway, enter search criteria, click **Find**, and choose a gateway from the resulting list.
- To add a new gateway:
  a. Click **Add New**.
  b. From the **Gateway Type** drop-down list, choose one of the supported gateway models.
  c. Click **Next**.

Step 3 Configure the MLPP fields on the **Gateway Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

Step 4 Click **Save**.

Configure Multilevel Precedence and Preemption for Phones



Caution Do not configure a device with the following combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.

Step 2 Enter search criteria.

Step 3 Click **Find** and choose a phone from the resulting list.

Step 4 Configure the MLPP fields on the **Phone Configuration** window. See the Related Topics section for more information about the fields and their configuration options.

Multilevel Precedence and Preemption Settings for Phones

Table 87: Multilevel Precedence and Preemption Settings for Phones

MLPP Settings for Phones Field	Description
Common Device Configuration	Choose the common device configuration that you configured. The common device configuration includes the attributes (services or features) that are associated with a particular user.
Calling Search Space	From the drop-down list, choose a calling search space (CSS) that you configured . A calling search space comprises a collection of partitions that are searched to determine how a dialed number should be routed. The calling search space for the device and the calling search space for the directory number are used together. The directory number CSS takes precedence over the device CSS.
MLPP Domain	Choose an MLPP domain from the drop-down list for the MLPP domain that is associated with this device. If you leave the **None** value, this device inherits its MLPP domain from the value that was set in the common device configuration. If the common device configuration does not have an MLPP domain setting, this device inherits its MLPP domain from the value that was set for the MLPP Domain Identifier enterprise parameter.
MLPP Indication	If available, this setting specifies whether a device that can play precedence tones will use the capability when it places an MLPP precedence call. From the drop-down list, choose a setting to assign to this device from the following options: • **Default**—This device inherits its MLPP indication setting from the common device configuration. • **Off**—This device does not handle nor process indication of an MLPP precedence call. • **On**—This device handles and processes indication of an MLPP precedence call. **Note** Do not configure a device with the following combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful. Turning on MLPP Indication (at the enterprise parameter or device level) disables normal Ring Setting behavior for the lines on a device, unless MLPP Indication is turned off (overridden) for the device.

MLPP Settings for Phones Field	Description
MLPP Preemption	Be aware that this setting is not available on all devices. If available, this setting specifies whether a device that can preempt calls in progress will use the capability when it places an MLPP precedence call. From the drop-down list, choose a setting to assign to this device from the following options: • **Default** —This device inherits its MLPP preemption setting from the common device configuration. • **Disabled**—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls. • **Forceful**—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.

Configure a Directory Number to Place Multilevel Precedence and Preemption Calls

After you configure a device, you can add a line (directory number) from the updated **Device Configuration** window.

Procedure

Step 1 From Cisco Unified CM Administration in the **Device Configuration** window, click **Add a new DN** for the appropriate line.

Step 2 In the **Target (Destination)** field, enter the number to which MLPP precedence calls should be diverted if this directory number receives a precedence call and neither this number nor its call forward destination answers the precedence call.

Values can include numeric characters, octothorpe (#), and asterisk (*).

Step 3 From the **MLPP Calling Search Space** drop-down list, choose the calling search space to associate with the MLPP alternate party target (destination) number.

Step 4 In the **MLPP No Answer Ring Duration (seconds)**, enter the number of seconds (between 4 and 60) after which an MLPP precedence call is directed to this directory number alternate party if this directory number and its call-forwarding destination have not answered the precedence call.

Leave this setting blank to use the value that is set in the **Precedence Alternate Party Timeout** enterprise parameter.

Step 5 Click **Save**.

Configure a User Device Profile for Multilevel Precedence and Preemption

When a user profile is assigned to a phone, the phone inherits the configuration of the assigned user, including any CSS that is associated with the user. The phone CSS can, however, override the user profile. Cisco Unified Communications Manager assigns the precedence level that is associated with the dialed pattern to the call when a pattern match occurs. The system sets the call request as a precedence call with the assigned precedence level.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Device Profile**.

Step 2 Perform one of the following tasks:

- To modify the settings for an existing device profile, enter search criteria, click **Find**, and then choose an existing device profile from the resulting list.
- To add a new device profile:
  - Click **Add New**.
  - From the **Device Profile Type** drop-down list, choose a profile type.
  - Click **Next**.
  - From the **Device Protocol** drop-down list, choose either **SIP** or **SCCP**.

Step 3 Click **Next**.

Step 4 From the **MLPP Domain** drop-down list, choose an MLLP domain that you configured.

Step 5 From the **MLPP Indication** drop-down list, choose one of the following settings to specify whether a device that is capable of playing precedence tones will use the capability when it places an MLPP precedence call:

- • **Default**—This device inherits its MLPP indication setting from its device pool.
  - **Off**—This device does not handle nor process indication of an MLPP precedence call.
  - **On**—This device does handle and process indication of an MLPP precedence call.

Step 6 From the **MLPP Preemption** drop-down list, choose one of the following settings to specify whether a device that is capable of preempting calls in progress will use the capability when it places an MLPP precedence call:

- • **Default**—This device inherits its MLPP preemption setting from its device pool.
  - **Disabled**—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.
  - **Forceful**—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.

Step 7 Click **Save**.

Configure the Default Device Profile for Multilevel Precedence and Preemption

Use the default device profile for whenever a user logs on to a phone model for which no user device profile exists. A default device profile comprises the set of services and features that are associated with a particular device.

Caution Do not configure a default device profile with the following combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.

Procedure

Step 1 In Cisco Unified CM Administration, choose **Device** > **Device Settings** > **Default Device Profile**.

Step 2 Perform one of the following tasks:

- To modify the settings for an existing default device profile, choose an existing default device profile from the **Device Profile Defaults** section.
- To add a new default device profile, choose a device profile type from the drop-down list, click **Next**, choose a device protocol, and then click **Next**.

Step 3 From the **MLPP Domain** drop-down list, choose an MLPP domain that you configured to associate to the device.

Step 4 From the **MLPP Indication** drop-down list, choose one of the following settings to specify whether a device that is capable of playing precedence tones will use the capability when it places an MLPP precedence call:

- **Default**—This device inherits its MLPP indication setting from its device pool.
- **Off**—This device does not handle nor process indication of an MLPP precedence call.
- **On**—This device does handle and process indication of an MLPP precedence call.

Step 5 From the **MLPP Preemption** drop-down list, choose one of the following settings to specify whether a device that is capable of preempting calls in progress will use the capability when it places an MLPP precedence call:

- **Default**—This device inherits its MLPP preemption setting from its device pool.
- **Disabled**—This device does not allow preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.
- **Forceful**—This device allows preemption of lower precedence calls to take place when necessary for completion of higher precedence calls.

Step 6 Click **Save**.

Multilevel Precedence and Preemption Interactions

Table 88: Multilevel Precedence and Preemption Interactions

Feature	Interaction
729 Annex A	729 Annex A is supported.
Cisco Extension Mobility	The MLPP service domain remains associated with a user device profile when a user logs in to a device by using extension mobility. The MLPP Indication and Preemption settings also propagate with extension mobility. If either the device or the device profile do not support MLPP, these settings do not propagate.
Cisco Unified Communications Manager Assistant	MLPP interacts with Cisco Unified Communications Manager Assistant as follows: • When Cisco Unified Communications Manager Assistant handles an MLPP precedence call, Cisco Unified Communications Manager Assistant preserves call precedence. • Cisco Unified Communications Manager Assistant filters MLPP precedence calls in the same manner as it filters all other calls. The precedence of a call does not affect whether the call is filtered. • Because Cisco Unified Communications Manager Assistant does not register the precedence of a call, it does not provide any additional indication of the precedence of a call on the assistant console.
Immediate Divert	Immediate Divert diverts calls to voice-messaging mail boxes regardless of the type of call (for example, a precedence call). When Alternate Party Diversion (call precedence) is activated, Call Forward No Answer (CFNA) is also deactivated.
Resource Reservation Protocol (RSVP)	RSVP supports MLPP inherently. The Cisco Unified Communications Manager System Guide explains how MLPP functions when RSVP is activated.
Supplementary Services	MLPP interacts with multiple line appearances, call transfer, call forwarding, three-way calling, call pickup, and hunt pilots as documented in the and the subsections that describe the interaction with each service.

Multilevel Precedence and Preemption Restrictions

Table 89: Multilevel Precedence and Preemption Restrictions

Restriction	Description
Bandwidth	Cisco Unified Communications Manager preempts lower precedence calls when adjusting video bandwidth for high priority calls. If the bandwidth is not sufficient to preempt, Cisco Unified Communications Manager instructs endpoints to use previously reserved lower video bandwidth. When Cisco Unified Communications Manager preempts a video call, the preempted party receives a preemption tone and the call gets cleared.
Call Detail Records	For the DRSN, CDRs represent precedence levels with values 0, 1, 2, 3, and 4 where 0 specifies Executive Override and 4 specifies Routine, as used in DSN. CDRs thus do not use the DRSN format.
Common Network Facility Preemption	Common Network Facility Preemption support exists only for T1-CAS and T1-PRI (North American) interfaces on targeted Voice over IP gateways that Cisco Unified Communications Manager controls by using MGCP protocol and that have been configured as MLPP Preemption Enabled.
Intercluster trunks	Intercluster trunk MLPP carries precedence information through dialed digits. Domain information does not get preserved and must be configured per trunk for incoming calls.
Line Groups	MLPP-enabled devices are not supported in line groups. We recommend the following guidelines: • MLPP-enabled devices should not be configured in a line group. Route groups, however, are supported. Both trunk selection and hunting methods are supported. • If an MLPP-enabled device is configured in a line group or route group, in the event of preemption, if the route list does not lock onto the device, the preempted call may be rerouted to other devices in the route/hunt list and preemption indication may be returned only after no devices are able to receive the call. • Route lists can be configured to support either of two algorithms of trunk selection and hunting for precedence calls. In method 1, perform a preemptive search directly. In method 2, first perform a friendly search. If this search is not successful, perform a preemptive search. Method 2 requires two iterations through devices in a route list. If route lists are configured for method 2, in certain scenarios involving line groups, route lists may seem to iterate through the devices twice for precedence calls.
Look Ahead For Busy	Cisco Unified Communications Manager does not support the Look Ahead for Busy (LFB) option.

Restriction	Description
MLPP Notification	Only MLPP Indication Enabled devices generate MLPP-related notifications, such as tones and ringers. If a precedence call terminates at a device that is not MLPP Indication Enabled, no precedence ringer gets applied. If a precedence call originates from a device that is not MLPP Indication Enabled, no precedence ringback tone is applied. If a device that is not MLPP Indication Enabled is involved in a call that is preempted (that is, the other side of the call initiated preemption), no preemption tone gets applied to the device. **Note** Before Release 15SU3, MLPP notifications did not appear on call-originating SIP endpoints until the call was connected when using a route or translation pattern. Starting with Release 15SU3, Unified CM supports MLPP notifications on specific SIP phone models. This feature enables these models to play precedence ringback notifications before the call connects. The supported models are the Cisco Desk Phone 9841, Cisco Desk Phone 9851, Cisco Desk Phone 9861, Cisco Desk Phone 9871, and Cisco Video Phone 8875 (including Non-Radio models).
Phones and trunks	For phones, devices that are MLPP indication disabled (that is, MLPP Indication is set to Off) cannot be preempted. For trunks, MLPP indication and preemption function independently.
Ring Setting Behavior	Turning on MLPP Indication (at the enterprise parameter, common device configuration, or device level) disables normal Ring Setting behavior for the lines on a device, unless MLPP Indication is turned off (overridden) for the device.
SCCP	IOS gateways support the SCCP interface to Cisco Unified Communications Manager. They support BRI and analog phones that appear on Cisco Unified Communications Manager as supported phone models. SCCP phones support the MLPP feature, and so do some phones with specific SIP loads. See the relevant phone administration and user guides for Cisco IP phone support information.

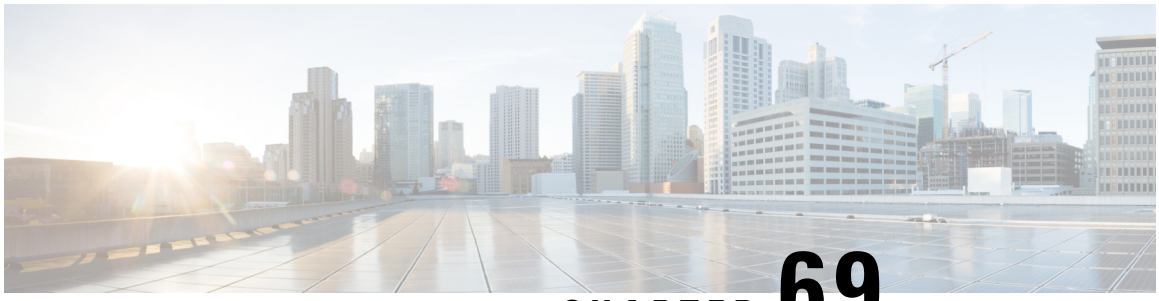
Restriction	Description
Supplementary Services	MLPP support for supplementary services specifies the following restrictions: • MLPP addresses only the basic Call Pickup feature and Group Call Pickup feature, not Other Group Pickup. • Call Forward All (CFA) support for inbound MLPP calls always forwards the call to the MLPP Alternate Party (MAP) target of the called party, if the MAP target is configured. In the event of an incorrect configuration (that is, if no MAP target is specified), the call is rejected, and the calling party receives a reorder tone. • Call Forward No Answer (CFNA) support for inbound MLPP calls forwards the call once to a CFNA target. After the first hop, if the call is unanswered, the call is sent to the MAP target of the original called party, if the MAP target has been configured. In the event of an incorrect configuration (that is, if no MAP target is specified), the call gets rejected, and the calling party receives a reorder tone . • Call Forward Busy (CFB) support for inbound MLPP calls forwards the call up to the maximum number that has been configured for forwarding hops. If the maximum hop count gets reached, the call gets sent to the MAP target of the original called party, if the MAP target has been configured. In the event of an incorrect configuration (that is, no MAP target is specified), the call gets rejected, and the calling party receives a reorder tone. • For hunt pilot support, the hunt group algorithm must specify Longest Idle Time, Top Down, or Circular. Ensure the hunt group options for busy treatment, no answer treatment, and unregistered treatment are set to Try next member, but do not go to the next group. Preemption only occurs across a single hunt group.
User Access Channel	User Access Channel support exists only for the following Cisco Unified IP Phone models, which must be configured as MLPP Preemption Enabled: • Cisco Unified IP Phone 7960, 7962, 7965 • Cisco Unified IP Phone 7940, 7942, 7945



PART XVI

SIP Interoperability

- Configure SIP Normalization and Transparency, on page 847
- Configure SDP Transparency Profiles, on page 853
- Configure Presentation Sharing using BFCP, on page 855
- Video Telephony, on page 859



CHAPTER 69

Configure SIP Normalization and Transparency

- SIP Normalization and Transparency Overview, on page 847
- SIP Normalization and Transparency Prerequisites, on page 848
- SIP Normalization and Transparency Configuration Task Flow, on page 849

SIP Normalization and Transparency Overview

SIP normalization and transparency is an optional feature that handles SIP interoperability issues between Unified Communications Manager and endpoints, service providers, PBXs, or gateways that implement SIP differently. To configure SIP normalization and transparency, apply a customized LUA script to a SIP trunk or SIP line. Unified Communications Manager applies the script to the SIP messaging that passes through the SIP trunks or SIP lines.

Upon installation, Unified Communications Manager contains default normalization and transparency scripts that you can assign to the SIP trunks and SIP profiles in your system. You can also create and import your own customized scripts.

SIP Normalization

SIP normalization scripts modify incoming and outgoing SIP messages. For example, if you are interoperating Unified Communications Manager with a Cisco TelePresence Video Communications Server, apply the *vcs-interop* script to the SIP trunk that connects the two. The script resolves the differences in the SIP messaging so that the two products can communicate.

You can apply a normalization script to any SIP trunk connection, regardless of which protocol is being used by the endpoint that connects to that SIP trunk.

SIP Transparency

SIP transparency scripts enable Unified Communications Manager to transparently pass SIP information, such as proprietary headers, from one call leg to the other. For transparency to work, both call legs must be SIP.

Another feature of SIP transparency is REFER transparency, which allows Unified Communications Manager to pass on REFER requests without acting on them. You can use REFER transparency in call center environments where a centralized agent may answer a call and then transfer the call to an agent who resides in the same geographical area as the caller. REFER transparency allows the centralized Unified Communications Manager to drop the call and shift call control to the new agent.

Default Scripts for SIP Normalization and Transparency

Upon installation, Cisco Unified Communications Manager contains the following default scripts for SIP Normalization and Transparency. You can apply these scripts to a SIP trunk or SIP profile, but you cannot edit these scripts. If none of these scripts meet your needs, you can create your own scripts:

- **cisco-meeting-server-interop**—Provides interoperability between Cisco Unified Communications Manager and Cisco Meeting Server (CMS).
- **cisco-telepresence-conductor-interop**—Provides interoperability for endpoints that are registered to TelePresence Conductor.
- **cisco-telepresence-mcu-ts-direct-interop**—Provides interoperability between Cisco Unified Communications Manager and either Cisco TelePresence MCU or Cisco TelePresence Server.
- **diversion-counter**—Provides capability to adjust the diversion counter.
- **HCS-PCV-PAI passthrough**—Provides Cisco HCS platform integration with Enterprise IMS.
- **redsky-alternate-id-interop**—Adds Redsky headers in outbound invite.
- **refer-passthrough**—Removes Cisco Unified Communications Manager from the call due to a blind transfer between SIP trunks.
- **vcs-interop**—Provides interoperability for endpoints that are registered to the Cisco TelePresence Video Communications Server.

SIP Normalization and Transparency Prerequisites

- Cisco Unified Communications Manager provides default scripts for SIP Normalization and Transparency. Make sure to review the existing scripts and system settings to verify whether they meet your needs. For information on the available default scripts, see Default Scripts for SIP Normalization and Transparency, on page 848.
- Make sure that you understand your deployment's SIP requirements in addition to the SIP requirements for any third-party products. For information on Cisco Unified Communications Manager's implementation of SIP, review the *SIP Line Messaging Guide for Cisco Unified Communications Manager (Standard Edition)* at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-programming-reference-guides-list.html.
- If you plan to develop customized SIP Normalization scripts, review the *Developer Guide for SIP Normalization and Transparency* at http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-programming-reference-guides-list.html.

SIP Normalization and Transparency Configuration Task Flow

Procedure

	Command or Action	Purpose
Step 1	Create New SIP Normalization and Transparency Scripts, on page 849	Optional. If none of the preinstalled scripts meet your needs, use this procedure to configure a customized script. You can create your new script in the **SIP Normalization Script Configuration** window or you can import a customized script.
Step 2	Apply Normalization or Transparency Script to SIP Trunk, on page 850	In the Trunk Configuration window, apply a script directly to a SIP trunk. Cisco Unified Communications Manager applies the script to all the SIP messaging that passes through the trunk
Step 3	Apply Normalization or Transparency to SIP Devices, on page 851	If you want to apply a normalization or transparency script to a SIP line, apply a script to the SIP profile that is associated to that SIP line. Cisco Unified Communications Manager applies the script to all SIP messaging that uses that SIP profile.

Create New SIP Normalization and Transparency Scripts

If the default normalization and transparency scripts do not meet your needs, use this procedure to create a new LUA script. You can either write the new script in Cisco Unified Communications Manager or import a file into the system.

Tip If the script that you want to create closely resembles a default script, open the default script in the **SIP Normalization Script Configuration** window and copy the **Contents** text box. Create a new script and paste the contents into the **Contents** text box. You can then edit the content in the new script.

Note The memory utilization of the SIP Normalization Script is based on each trunk and not on each script.

Procedure

Step 1 In Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Normalization Script**.

Step 2 Click **Add New**.
The SIP Normalization Script Configuration window appears.

Step 3 Enter a **Name** and **Description** for your script.

Step 4 If you are writing a new script, edit the script in the **Contents** text box.

Step 5 Optional. If you have an external file that you want to import, do the following

a) Click **Import File**.
b) **Browse** to locate the file and select the file.
c) Click **Import File**.
The **SIP Normalization Script Configuration** window displays the contents of the imported file in the **Contents** text box.

Step 6 Complete the fields in the **SIP Normalization Script Configuration** window. For help with the fields and their contents, refer to the online help.

Step 7 Click **Save**.

What to do next

Assign the script to a SIP profile or SIP trunk:

- Apply Normalization or Transparency to SIP Devices, on page 851
- Apply Normalization or Transparency Script to SIP Trunk, on page 850

Apply Normalization or Transparency Script to SIP Trunk

Use this procedure to apply a SIP normalization or transparency script to a SIP trunk. Cisco Unified Communications Manager applies the script to all SIP messaging that passes through the trunk.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Trunk**.

Step 2 Click **Find** and select the trunk to which you want to apply a script.

Step 3 From the **Normalization Script** drop-down list, choose the script that you want to apply to the trunk.

Step 4 (Optional) If you want to normalize specific parameters within the SIP messaging, do the following:

a) Enter the **Parameter Name** that you want to normalize, and the **Parameter Value** for the value that you want to apply to the parameter. For example, you could enter a `Location` parameter and `North Carolina` as the value.
b) To add additional parameters, click the **(+)** to create additional lines where you can enter additional parameters and values.

Step 5 (Optional) If you want to produce SDI traces against the script, check the **Enable Trace** check box.

Note
Cisco recommends that you enable tracing while debugging your scripts.

Step 6 Click **Save**.

Apply Normalization or Transparency to SIP Devices

You can apply a customized SIP Normalization and Transparency script, or a customized SDP Transparency Profile to a SIP phone by applying the script to the SIP Profile that is used by that device.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.

Step 2 Click **Find** and select the SIP profile to which you want to apply a script.

Step 3 In the **SDP Information** area, from the **SDP Transparency Profile** drop-down list, choose a profile.

Step 4 From the **Normalization Script** drop-down list, choose the script that you want to apply to the trunk.

Step 5 (Optional) If you want to normalize specific parameters within the SIP messaging, do the following:

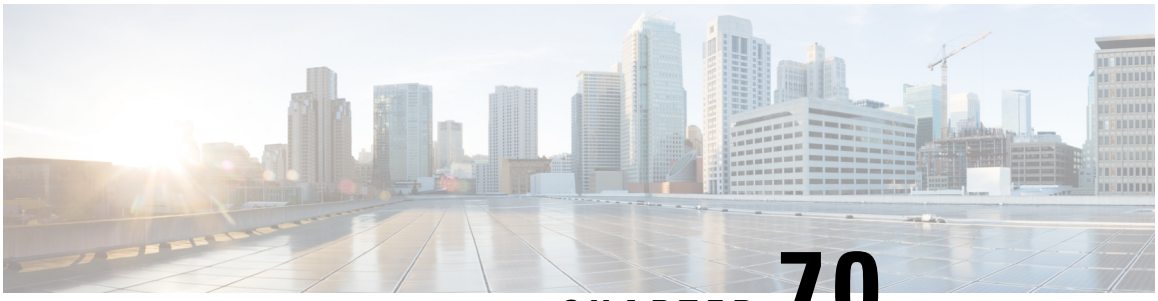
a) Enter the **Parameter Name** that you want to normalize, and the **Parameter Value** for the value that you want to apply to the parameter. For example, you could enter a **`Location`** parameter and **`North Carolina`** as the value.

b) To add additional parameters, click the **(+)** to create additional lines where you can enter additional parameters and values.

Step 6 (Optional) If you want to produce SDI traces against the script, check the **Enable Trace** check box.

Note
Cisco recommends that you enable tracing while debugging your scripts.

Step 7 Click **Save**.



CHAPTER 70

Configure SDP Transparency Profiles

- SDP Transparency Profile Overview, on page 853
- SDP Transparency Profile Restrictions, on page 853
- SDP Transparency Profile Prerequisites, on page 854
- Configure SDP Transparency Profile, on page 854

SDP Transparency Profile Overview

SDP Transparency Profiles contain a set of rules for declarative SDP attributes that allow the system to pass through declarative attributes that are not natively supported by Unified Communications Manager from the ingress to the egress call leg. Without an SDP transparency profile, Unified Communications Manager drops non-supported SDP attributes.

You can configure SDP transparency profiles with multiple rules and apply them to SIP devices via the SIP profile. In order for the SDP transparency profile to be applied, both call legs must be SIP. You can configure the following types of rules for SDP attributes:

- Property—If a rule is configured for a property attribute, Unified Communications Manager passes through the SDP attribute unless the attribute has a value.
- Any Value—If a rule is configured for any value, the SDP attribute gets passed through so long as it has a value that consists of at least one non-white space character.
- Value From List—If a rule is configured using this option, the SDP attribute gets passed through so long as it matches one of the specified values. You can configure up to five possible values

SDP Transparency Profile Restrictions

The following restrictions apply to SDP transparency profiles. If any of these situations occur on the egress call leg, Cisco Unified Communications Manager will not pass through the declarative SDP attribute:

- One or more Media Termination Points (MTPs) or Trusted Relay Points (TRPs) that do not support passthrough are allocated
- The Media Termination Point Required check box is checked for the SIP trunk
- A transcoder is being used
- RSVP is being used

- The ingress call leg is using Delayed Offer while the egress call leg is using Early Offer
- The media line has been rejected (port=0)
- Either call leg is using a protocol other than SIP

SDP Transparency Profile Prerequisites

If you plan to deploy any third-party SIP products, make sure that you understand how the products implement the Session Description Protocol (SDP).

Configure SDP Transparency Profile

Configure a customized SDP Transparency Profile with a set of rules for declarative SDP attributes that are not natively supported by Cisco Unified Communications Manager.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SDP Transparency Profile**.

Step 2 Click **Add New**.

Step 3 Enter a **Name** and **Description**.

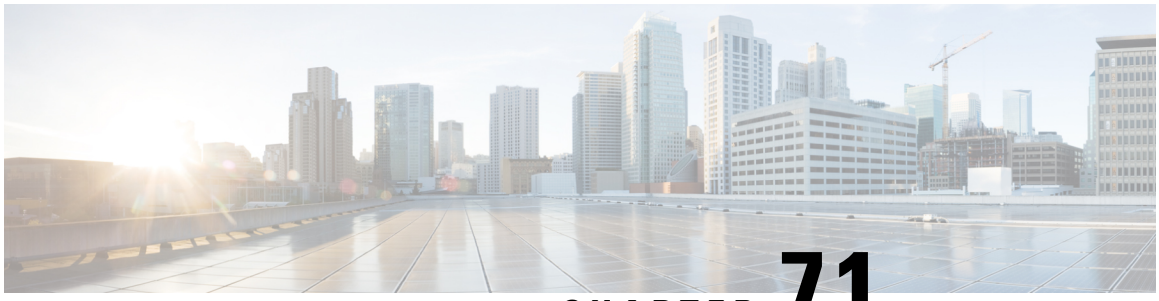
Step 4 In the **Attribute Information** pane, create the rules for the SDP attributes that you want to pass through:

- To pass through a property attribute, enter the attribute in the **Name** text box (for example, a=recvonly) and from the **Type** drop-down list, select **Property**.
- To pass through a value attribute, enter the attribute in the **Name** text box (for example, a=rtpmap), and select **Any Value** from the **Type** drop-down list box.
- To pass through a value attribute with any of up to five values, enter the attribute in the **Name** field (for example, a=rtpmap) and select **Any Value** from the **Type** drop-down list. In the resulting **Value** text box, enter the value of the attribute. You can click **+** to add up to five possible values for this attribute.

Step 5 Click the **(+)** to create new lines where you can enter additional SDP attributes for this transparency profile.

Step 6 Click **Save**.

Note
You must apply this profile to a SIP Profile so that the devices that use the SIP Profile can use the SDP Transparency Profile.



CHAPTER 71

Configure Presentation Sharing using BFCP

- Binary Floor Control Protocol Overview, on page 855
- Presentation Sharing using BFCP Prerequisites, on page 856
- Presentation Sharing using BFCP Configuration Task Flow, on page 857

Binary Floor Control Protocol Overview

Unified Communications Manager supports presentation sharing using the Binary Floor Control Protocol (BFCP) for supported Cisco endpoints and third-party video endpoints. This feature lets users share a presentation within ongoing audio or video conversation.

The following example describes how presentation sharing works using BFCP:

- An ongoing video conversation exists between two video phones. User A decides to share content with User B during the conversation. User A has the option to share the entire screen or share the specific application.
- The BFCP stream allows User B to view User A's shared content.

An audio-video call with content share requires at least four channels: audio, main video, the second video, and BFCP control channel, to achieve video conferencing and sharing presentations in the second video channel. If the call parties are capable of Far-End Camera Control (FECC), a fifth channel must be established.

Presentation Sharing with BFCP

From release 12.5(1)SU3 onwards, for Unified Communications Manager registered SIP endpoints, the BFCP work when:

- Two video-capable endpoints that start the conversation in audio-only mode share content during the call using BFCP support.
- TRP is allocated during the call.

BFCP Architecture

Presentation sharing using BFCP is supported only on BFCP enabled SIP networks. The entire network, including the endpoint devices and trunks, must be SIP.

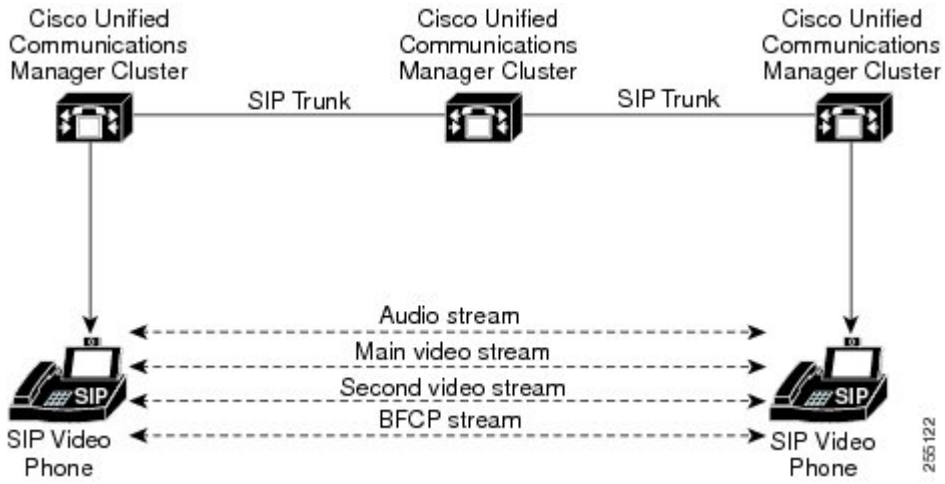
Unified Communications Manager aids in the negotiation of the BFCP stream by relaying SIP messages between two endpoints.

This negotiation involves establishing a floor, which is a temporary permission to access shared resources.

The BFCP stream is a point-to-point stream between the endpoints. Unified Communications Manager is never a target of the BFCP stream.

The following figure provides an example of a complex video network with multiple Unified Communications Manager clusters. BFCP must be enabled on all the trunks and lines connecting the devices. For this network, BFCP must be enabled on the four SIP trunks and two SIP lines that connect the endpoints.

Figure 14: Video Network with Multiple Cisco Unified Communications Manager Clusters



BFCP Limitations

Unified Communications Manager rejects the BFCP stream in the following scenarios:

- The **Allow Presentation Sharing using BFCP** check box on the SIP Profile page is unchecked for one of the SIP lines or trunks in the network.
- One endpoint offers BFCP, but the other does not.
- When the SIP line or SIP trunk uses MTP (non pass-through mode) or Transcoder.

Note BFCP control channel is always unencrypted. However, the presentation channel is encrypted if both phones are encrypted.

Presentation Sharing using BFCP Prerequisites

- Make sure all endpoints and trunks in the call flow are running SIP profile.
- Check Phone Support procedure and generate a report for the feature **BFCP Support** to obtain a list of Cisco endpoints that support Presentation Sharing using BFCP. For these endpoints, BFCP support is

enabled by default. You need not perform any additional configuration for the phone to support BFCP. For more information, see Generate a Phone Feature List, on page 5.

Presentation Sharing using BFCP Configuration Task Flow

Complete the following tasks to enable Presentation sharing using the Binary Floor Control Protocol (BFCP).

Procedure

	Command or Action	Purpose
Step 1	Enable BFCP Support for SIP Trunks, on page 857	Enables BFCP support on all SIP trunks in the call flow.
Step 2	Enable Presentation Sharing using BFCP for Third-Party Phones, on page 858	Enables BFCP support in the third-party phone configuration, if you are using third-party SIP endpoints.

Enable BFCP Support for SIP Trunks

If you are using Presentation sharing with BFCP, the feature must be enabled in the SIP Profile that is used by all trunks in the messaging or call flow. The BFCP stream will be rejected by any trunk that does not support presentation sharing.

Procedure

Step 1 Enable BFCP support within the SIP Profile that is used by the SIP trunk:

a) From the Cisco Unified CM Administration, choose **Device** > **Device Settings** > **SIP Profile**.

b) Perform either of the following steps:

- Click **Find** to select an existing SIP profile.
- Click **Add New** to create a new SIP profile.

c) In the **SDP Information** section, check the **Allow Presentation Sharing using BFCP** check box to enable BFCP in the Unified Communications Manager.

By default, the check box is unchecked. For presentation sharing, BFCP must be enabled for all SIP trunks between the Unified CM clusters.

d) Complete any other fields in the **SIP Profile Configuration** window. For more information on the fields and their configuration options, see the system Online Help.

e) Click **Save**.

Step 2 Associate the BFCP enabled SIP Profile to your SIP trunks:

a) From the Cisco Unified CM Administration, choose **Device** > **Trunks**.

b) Click **Find** and select an existing SIP trunk.

c) In the **SIP Information** section, choose the SIP Profile for which you enabled BFCP to share the presentation in the intercluster call from the **SIP Profile** drop-down list.
d) Click **Save**.
e) Repeat this step for all SIP trunks that will be in the call flow of a BFCP session.

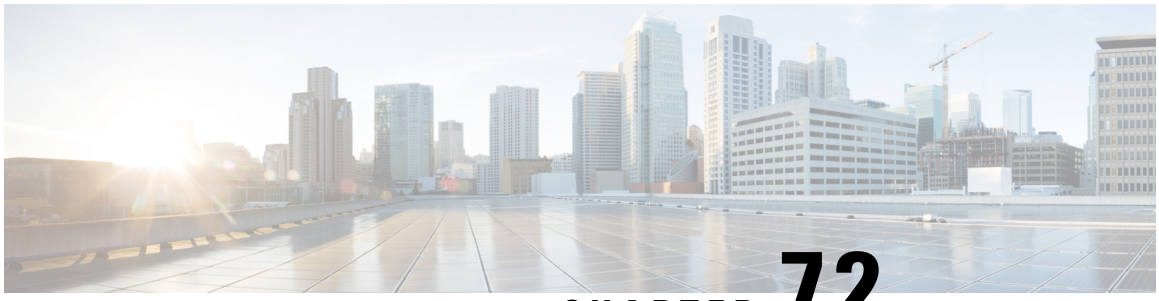
Enable Presentation Sharing using BFCP for Third-Party Phones

If you want to use Presentation sharing using BFCP with third-party SIP Phones, you must make sure that the feature is enabled for the endpoint. This feature is supported by the following third-party phone types:

- Third-party SIP Device (Advanced)
- Third-party AS-SIP Endpoint

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Phone**.
Step 2 Click **Find** and select an existing third-party SIP phone.
Step 3 Check the **Allow Presentation Sharing using BFCP** check box.
Step 4 Click **Save**.



CHAPTER 72

Video Telephony

- Video Telephony Overview, on page 859
- Video Telephony Support, on page 859
- Video Network, on page 863
- Video Telephony Configuration Task Flow, on page 865
- H.323 Video, on page 865
- Video Support, on page 870
- Video Features , on page 873
- QoS for Video Networks , on page 875

Video Telephony Overview

Unified Communications Manager supports video telephony and thus unifies the world of voice and video calls. Video endpoints use Unified CM call-handling features and access a unified voice and video solution for dialing and connecting video calls.

The Unified Communications Manager video telephony solution offers these features:

- Supports video and video-related features, such as Far-End Camera Control (FECC)
- Supports multiple logical channels that are needed to allow the transmission of video streams
- Transmits midcall, media-related messages that are needed for video (that is, transmits commands or indications that are needed for video calls)
- Supports H.323, Skinny Client Control Protocol (SCCP), and Session Initiation Protocol (SIP)
- Enhances locations and regions to provide Bandwidth Management
- Provides serviceability information, such as Call Detail Records (CDRs), about video calls

Video Telephony Support

The following sections discuss the details of video telephony in the Unified Communications Manager environment.

Video Calls

The typical video call includes two or three Real-Time Protocol (RTP) streams in each direction (that is, four or six streams). The call can include the following stream types:

- Video (H.261, H.263, H.263+, H.264-SVC, X-H.264UC, H.264-AVC, H.265, AV1 and VT Camera wideband video codecs)
- Far-End Camera Control (FECC) - Optional
- Binary Floor Control Protocols (BFCP)

Note Call control for video calls operates the same way as the call control that governs all other calls. For more information, see the Configure Media Resources chapter in the System Configuration Guide. You can also see, Configure Conference Bridges chapter in the System Configuration Guide for details on how Unified Communications Manager can allocate a video conference bridge automatically.

Real-Time Transport Control Protocol Pass-Through in MTP Topologies

An IOS Media Terminate Point (MTP) before 15.2(2)T, cannot pass-through Real-Time Transport Control Protocol (RTCP) packets and therefore cannot exchange Real-Time Protocol (RTP) feedback data to enhance the RTP transmission. The primary function of RTCP is to provide feedback on media distribution by periodically sending statistics to participants in a streaming multimedia session. RTCP gathers statistics for a media connection and information such as transmitted octet and packet counts, lost packet counts, jitter, and round-trip delay time. An application may use this information to control the quality of service parameters, perhaps by limiting flow or using a different codec.

IOS MTP Version 15.2(2)T and later supports RTCP pass-through capability so that the endpoint in a call with an MTP present can still provide feedback and status on an RTP transmission. RTCP pass-through capability applies to media channels.

The RTCP pass-through feature is not limited to a specific call signaling protocol. For example, it can be SIP-SIP, SIP-nonSIP, or nonSIP-nonSIP.

For Unified CM to allocate an RTCP pass-through capable MTP specifically, the call needs to fulfill the following conditions:

- An MTP is requested for a feature that requires the MTP to be in media pass-through mode. For example, TRP, DTMF translation, IP address V4/V6 translation, and so on. RTCP pass-through is only applicable when the media is in pass-through mode.
- The RTCP pass-through MTP needs to be included in the Media Resource Group Lists (MRGL) of the endpoint that sponsors the MTP. MTP can be inserted by RSVP, TRP, DTMF mismatch reasons.
- When the call is capable of establishing video channels, Unified CM attempts to search for an RTCP pass-through capable MTP. For example, Unified CM picks an RTCP pass-through capable MTP from other non-capable ones in the MRGL. If an RTCP pass-through capable MTP is not available, then Unified CM stills allocate an MTP for the call.
- When the call is capable of establishing an audio channel only, Unified CM does not intentionally request an RTCPpass-through capable MTP for the non-video calls. However, if the MRGL only contains RTCPpass-through capable MTP(s), then Unified CM inserts one of those into the audio call.

- The call also needs to fulfill the current CAC bandwidth for video calls to have an RTCP pass-through capable MTP.

Note If a call initially establishes with a non-RTCP pass-through capable MTP (before version 15.2(2)T) present in the call, and the call escalates into a video-capable call, Unified CM does not reallocate to an RTCP pass-through capable MTP. In that case, even though the call has been escalated to a video call, the existing MTP does not allow RTCP packets to be passed through.

Video Codecs

Common video codecs include H.261, an older video codec, H.263, a newer codec that gets used to provide internet protocol (IP) video, and H.264, a high-quality codec. The system supports H.264 for calls that use the Skinny Client Control Protocol (SCCP), H.323, and SIP on originating and terminating endpoints only. The system also supports regions and locations.

Unified Communications Manager maintains the offerer's video codec ordering preference when making the answer, if possible. H.265 is the preferred video codec is available on the endpoints, otherwise, Unified Communications Manager follows the following codec preference preference order:

Preference Order	Codecs	Description
1	H.265 (HEVC)	Provides higher quality video using lower bandwidth.
2	H.264 (SVC)	Allows rendering of variable quality video from the same media stream, by disregarding a subset of the packets received. **Note** H. 264 SVC is a new annex to the H.264-AVC video compression standard; meaning it is an enhancement on top of H.264-AVC. It provides the ability to encapsulate multiple video streams at various frame-rates and resolutions in one container.
3	X-H.264UC (Lync)	Microsoft-Proprietary Variant
4	H.264 (AVC)	Advanced Video Coding

Preference Order	Codecs	Description

5	H.263	H.263 and H.261 codecs exhibit the following parameters and typical values:
		• Bit rates range from 64 kb/s to a few mb/s. These bit rates can exist in any multiple of 100 b/s. H.261 and H.263 can function with bit rates lower than 64 kb/s, but video quality suffers in such cases.
		  • One-quarter Common Interchange Format (QCIF) (Resolution equals 176x144.)
		  • Common Interchange Format (CIF) (Resolution equals 352x288.)
		  • 4CIF (Resolution equals 704x576.)
		  • Sub QCIF (SQCIF) (Resolution equals 128x96.)
		  • 16CIF (Resolution equals 1408x1152.)
		  • Custom Picture Format
		• Resolution:
		• Frame Rate: 15 frames per second (fps), 30 fps
		• Annexes: F, D, I, J, K, L, P, T, N

6	H.261	

The bandwidth of video calls equals the sum of the audio bandwidth and the video bandwidth. The total bandwidth does not include overhead.

A 384-kb/s video call may comprise G.711 at 64 kb/s (for audio) plus 320 kb/s (for video). This sum does not include overhead. If the audio codec for a video call is G.729 (at 24 kb/s), the video rate increases to maintain a total bandwidth of 384 kb/s. If the call involves an H.323 endpoint, the H.323 endpoint may use less than the total available video bandwidth. Regardless of protocol, the endpoint may always choose to send at less than the max bit rate for the call.

AV1 Codec Support

AV1 is a next-generation video codec developed by the Alliance for Open Media. The benefits of AV1 are:

- Reduced bandwidth consumption and better visual quality by utilizing better compression efficiency compared to other video encodings
- Enables video for users on very low bandwidth networks
- Significant screen sharing efficiency improvements over other codecs

Unified Communication Manager supports negotiation of AV1 codec to establish media if endpoints support the AV1 codec.

When both endpoints support Multiple Codecs in Answer, Unified CM negotiates all the matching codecs including AV1 based on the preference order received. The endpoint will then use one of the codecs from the

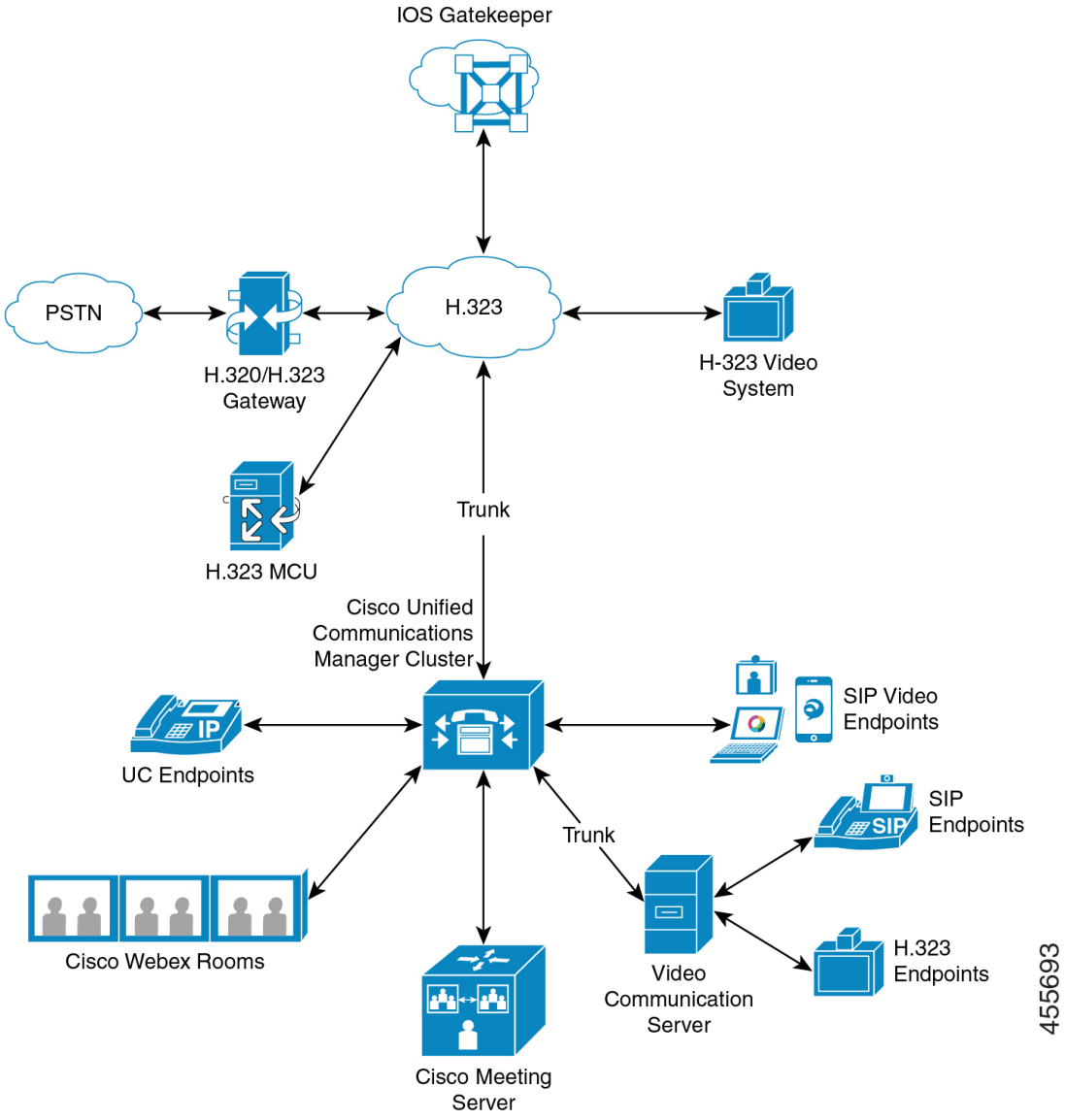
negotiated codec list for media streaming. In a low bandwidth environment, the AV1 codec is preferred by the endpoint over other codecs in the negotiated list.

When both the endpoints involved in the call do not support the Multiple Codec in Answer, and the AV1 is the preferred codec over other codecs, Unified CM selects AV1 as the negotiated codec.

Note AV1 codec is not yet supported on TelePresence endpoints.

Video Network

The following illustration provides an example of a video network that uses a single Unified Communications Manager cluster. In a successful video network, any endpoint can call any other endpoint. Video availability only exists if both endpoints are video-enabled. Video capabilities extend across trunks.



The Cisco video conference portfolio comprises the following video bridges:

- Cisco TelePresence MCU series
- Webex Meeting Server

The Cisco UC Endpoints portfolio comprises the following endpoints that support video:

For more information on Cisco UC Endpoint portfolio that support vedio, see Compatibility Matrix.

Note Third-party SIP video endpoints are capable of connecting to Cisco Unified Communications Manager as a line-side device or as a trunk-side device. For more information, see Third-Party SIP Endpoints.

Video Telephony Configuration Task Flow

To configure video telephony in Cisco Unified Communications Manager Administration follow these steps.

Procedure

Step 1 Configure regions for video call bandwidth if you use regions for call admission control.

Note
All devices include a default region, which defaults to 384 kb/s for video. You can set the bandwidth setting in Region configuration high enough for the desired resolution (For example, increase to 2Mb/s for high-definition video call).

Step 2 Configure locations for video call bandwidth if you use locations for call admission control.

Step 3 (Optional) Configure the RSVP service parameters, or set the RSVP policy in the Location Configuration window if using RSVP for bandwidth management of SIP video calls.

Step 4 Configure the appropriate conference bridge for your network to use a Cisco video conference bridge.

Step 5 Configure the media resource groups and media resource group lists for the user accordingly to configure a user to use the video conference bridge instead of using other conference bridges.

Step 6 Configure the H.323 gateways in your system to retry video calls as audio calls (default behavior) or configure AAR groups and route/hunt lists to use alternate routing for video calls that do not connect.

Step 7 Configure the H.323 phones in your system to retry video calls as audio calls (default behavior) or configure AAR groups and route/hunt lists to use alternate routing for video calls that do not connect. Choose **Enabled for Video Capabilities**.

Step 8 Configure the H.323 trunks in your system to retry video calls as audio calls (default behavior) or configure AAR groups and route/hunt lists to use alternate routing for video calls that do not connect.

Step 9 Configure the Cisco Unified IP Phones that will support video.

Step 10 Configure the third-party SIP endpoints that will support video.

Step 11 Configure the SIP trunks in your system to retry video calls as audio calls (default behavior).

H.323 Video

H.323 video exhibits the following characteristics:

- H.323 endpoints can be configured as H.323 phones, H.323 gateways, or H.323 trunks.
- Call forwarding, dial plan, and other call-routing-related features work with H.323 endpoints.
- H.323 video endpoints cannot initiate hold, resume, transfer, park, and other similar features.
- If an H.323 endpoint supports the empty capability set (ECS), the endpoint can be held, parked, and so forth.

- Some vendors implement call setup in such a way that they cannot increase the bandwidth of a call when the call gets transferred or redirected. In such cases, if the initial call is audio, users may not receive video when they are transferred to a video endpoint.
- No video media termination point (MTP) nor video transcoder currently exists. If an audio transcoder or MTP is inserted into a call, that call will be audio only. This is true when the IPVC audio transcoding capabilities is not being used. When the IPVC transcoders are used, you can transcode the audio and send/receive video.
- For H.323 video calls, users must specify video call bandwidth.

H.239-Extended Video Channels in H.323 Call

The extended video channels feature works via H.239 protocol and enables multiple video channel support. Cisco Unified Communications Manager supports negotiating an extended video channel using the H.239 protocol in direct point-to-point H.323 calls. This also includes calls across the H.323 intercluster trunk.

Cisco Unified Communications Manager supports all H.239 associated support signals and commands that are specified in the H.239 recommendation.

The following sections describes characteristics which apply to the extended video channels feature.

Support for Third-Party H.323 Devices

The extended video channel feature supports H.239 interoperability among third-party video endpoints and Cisco Unified Voice Conferencing. Cisco Unified Communications Manager allows an extended video channel to be used for presentation and live meeting transmission. This feature focuses on multiple video channel support via H.245 signaling. The following presentation applications provide basis for this multichannel support:

- Natural Presenter package by the third-party vendor video endpoint
- People + Content by the third-party vendor Polycom

Both Natural Presenter package and People + Content use the H.239 protocol to negotiate capabilities and define the roles of the additional video channels.



Note Natural Presenter package by video endpoint and People + Content by Polycom only support H.239 for the presentation mode.

Be aware that the presentation applications that are offered by video endpoint and Polycom are optional features. You must have one of these options and H.239 enabled in both caller and callee endpoints to negotiate second video channels or the call will be limited to a single video channel.

H.323 Devices Invoke Presentation Feature

Cisco and Polycom video endpoints allow the user to share presentation materials from various components (for example, VCR, Projector, PC, and so on). The components can physically connect with the endpoints, and the PC can also run presentation applications that are provided by the vendor to transmit the presentation image. The source of presentation and the component connection with the video endpoint are irrelevant to the mechanism of establishing video channels by using H.239.

Note For details on setting up presentation sources, see the video endpoint user guide.

When two H.239-enabled endpoints attempt to establish a video call, they declare their video capabilities for the main video channel for meeting participants and their extended video capabilities (H.239 capabilities) for the second video channel. The following contents comprise H.239 capability signals:

1. The endpoints send signals to indicate that the devices support H.239. They also send associated commands and indication signals for managing the second video channel. This enables both the endpoints to be aware that the call is capable of opening multiple video channels.
2. The endpoints sends out one or more extended video codec capabilities to express video codec capabilities for second channels. The endpoint must specify the role of the second video channel. The defined role labels can be
    - Live-video-This channel gets processed normally and is suitable for live video of people
    - Presentation-This channel relays a token-managed presentation that is distributed to the devices

After the capabilities have been exchanged, both endpoints immediately open two-way audio channels and the main video channels as in the traditional video calls.

Opening Second Video Channels

Depending on the third-party endpoint implementation, the second video channel is handled differently among vendors.

Natural Presenter Package by Tandberg

Video endpoints initiates the second video channel on demand. A video endpoint device does not open the second video channel immediately after the main video channel is established. The second channel gets opened when one of the callers (the presenter) specifies the source of the presentation and invokes a command to start the presentation.

When a video endpoint user decides to start sharing the presentation, video endpoint requests the other call party to open an extended video channel for receiving the presentation image; therefore, a video endpoint-video endpoint call has only one-way second video channel.

People + Content by Polycom

Unlike video endpoint, a Polycom video endpoint initiates the second video endpoint immediately as a part of the default mechanism, after both parties have confirmed that additional video channels can be supported.

Note The channel established gets automatically if both parties support H.239 and have the extended video channel feature enabled. However, the additional channel does not show anything until one of the parties start to share presentation.

Polycom initiates a request for the second video channel to the other call party regardless of the usage of the second video channel; therefore, in a Polycom-Polycom call, two-way video channels get opened between the devices even if only one of them sends out presentation image/video.

This implementation ensures that both call parties have the second video channel ready for transmission when the call parties decide to take the token to present something. Although one of the two video channels remains idle (not sending anything), the Polycom device controls bandwidth to ensure load efficiency.

This difference in handling second video channels does not affect the implementation of H.239. Unified Communications Manager does not initiate any receiving channel request in an H.323-H.323 call. Unified Communications Manager simply relays all channel requests from one terminal to another.

Unified Communications Manager does not enforce two-way transmission for the second set of video channels because this does not represent a requirement in the H.239 protocol.

Call Admission Control (CAC) on Second Video Channels

The following call admission control policies of Cisco Unified Communications Manager get applied to the second video channels:

Cisco Unified Communications Manager restricts the bandwidth usage by the second video channels on the basis of location configuration. When the second video channel is being established, Cisco Unified Communications Manager makes sure that enough video bandwidth stays available within the location pool and reserves bandwidth accordingly. If the required bandwidth is not available, Cisco Unified Communications Manager instructs the channel to reduce the available bandwidth to zero.

No change occurs in the region configuration or policies to support the second video channels.

Traditionally, Cisco Unified Communications Manager region policy has only supported a call with a single video channel and the total bandwidth usage of this call never gets larger than what the region configuration specifies.

If the administrator sets a finite region video bandwidth restriction for an H.239 call, Cisco Unified Communications Manager will violate the region policy because the region value will get used against the bandwidth that is requested for each video channel independently.

Example

```
If the region video bandwidth is set to 384Kbps and the audio channel uses 64Kb/s,

the maximum allowed bandwidth for each video channel will be (384Kb/s - 64Kb/s)=
 320Kb/s.
i.e. the maximum bandwidth to be used by the H.239 call will be (audio bw + 2*(384
 - audio
bw)) = 704Kb/s, which goes beyond the 384Kb/s bandwidth that the region specifies.
```

Note You should consider relaxing both region and location bandwidth restrictions for H.239 calls, so the H.239 devices are allowed to readjust and balance load for both the video channels without Cisco Unified Communications Manager intervention.

Number of Video Channels Allowed

Unified Communications Manager supports only a maximum of two video channels due to the following reasons:

- Both Cisco and Polycom only support two video channels, one of which is for main video, and the other is for presentation.
- H.239 only defines an Additional Media Channel (AMC) for H.320-based system to partition the traditional H.320 video channel for the purpose of presentation.

H.239 Commands and Indication Messages

Command and Indication (C&I) messages get used for H.239 to manage tokens for the Presentation and Live roles and to permit devices to request release of video flow control to enable the operation of additional media channels. Cisco Unified Communications Manager supports all the C & I messages. Whenever Cisco Unified Communications Manager receives C&I messages, it relays them to the call party accordingly.

Be aware that the flow control release request and response messages can be used to request that the far end release flow control, so it allows an endpoint to send the indicated channel at the indicated bit rate.

Note Be aware that the call party may or may not honor the request as is indicated by flow control release response.

The Presentation role token messages allow an H.239 device to acquire the token for presentation. The other call party may accept or reject the request. The presenter device sends out a token release message when it is no longer needed.

Topology and Protocol Interoperability Limitation

Cisco Unified Communications Manager supports only H.239 in H.323 to H.323 calls. Cisco Unified Communications Manager allows H.239 calls to be established across H.323 intercluster trunk or multiple nodes. If an H.239-enabled device attempts to make a call with a non-H323 end, the H.239 capabilities will get ignored and the call will get conducted like the traditional video calls that are supported by Cisco Unified Communications Manager.

Cisco Unified Communications Manager does not support a second video channel when a media termination point or transcoder is inserted into the call. If it happens, the call will fall back to normal video calls.

Midcall Feature Limitation

Cisco Unified Communications Manager supports opening second video channels only in direct H.323 to H.323 calls.

Caution Do not attempt to invoke any midcall features such as call transfer or hold/resume operations. Doing so can lead to problems and the second video channel can get disconnected.

Video Support

Unified Communications Manager supports video over H.323, SCCP and SIP protocols.

Skinny Client Control Protocol Video

Skinny Client Control Protocol video exhibits the following characteristics:

- If a phone that is running Skinny Client Control Protocol reports video capabilities, Cisco Unified Communications Manager automatically opens a video channel if the other end supports video.
- For Skinny Client Control Protocol video calls, system administration determines video call bandwidth by using regions. The system does not ask users for bit rate.

SIP Video

SIP video supports the following video calls by using the SIP Signaling Interface (SSI):

- SIP to SIP
- SIP to H.323
- SIP to SCCP
- SIP intercluster trunk
- H.323 trunk
- Combination of SIP and H.323 trunk

SIP video calls also provide media control functions for video conferencing.

Unified Communications Manager video supports SIP on both SIP trunks and lines support video signaling. SIP supports the H.261, H.263, H.263+, H.264 (AVC), H.264 (SVC), X-H.264UC (Lync), and AV1 video codecs (it does not support the wideband video codec that the VTA uses).

Note Only some of the endpoints supports AV1 codec. For more information, see Compatability Matrix.

Configuring SIP Devices for Video Calls

Perform the following steps to enable video calls on SIP devices:

SIP Trunks

- On the Trunk Configuration window in Unified Communications Manager, check the **Retry Video Call** as **Audio** check box if you want the call to use audio when the video connection is not available.
- Reset the trunk.

Third-Party SIP Endpoints

- On the Phone Configuration window in Cisco Unified Communications Manager Administration, check the **Retry Video Call** as **Audio** check box if you want the call to use audio when the video connection is not available.
- Reset the endpoint.

Cisco Video Conference Bridges

Unified Communications Manager supports a variety of solutions for video conferencing. The following video conference bridges support ad hoc and meet-me video conferencing:

- Cisco TelePresence MCU
- Cisco TelePresence Conductor
- Cisco Meeting Server

Cisco TelePresence MCU Video Conference Bridge

Cisco TelePresence MCU is a set of hardware conference bridges for Cisco Unified Communications Manager.

The Cisco TelePresence MCU is a high-definition (HD) multipoint video conferencing bridge. It delivers up to 1080p at 30 frames per second, full continuous presence for all conferences, full transcoding, and is ideal for mixed HD endpoint environments. The Cisco TelePresence MCU supports SIP as the signaling call control protocol. It has a built in Web Server that allows for complete configuration, control, and monitoring of the system and conferences. The Cisco TelePresence MCU provides XML management API over HTTP.

Cisco TelePresence MCU allows both ad hoc and meet-me voice and video conferencing. Each conference bridge can host several simultaneous, multiparty conferences. Cisco TelePresence MCU must be configured in Port Reservation mode.

Cisco TelePresence Conductor Video Conference Bridge

Cisco TelePresence Conductor provides intelligent conference administrative controls and is scalable, supporting device clustering for load balancing across MCUs and multiple device availability. Administrators can implement the Cisco TelePresence Conductor as either an appliance or a virtualized application on VMware with support for Cisco Unified Computing System (Cisco UCS) platforms or third-party-based platforms. Multiway conferencing, that allows for dynamic two-way to three-way conferencing, is also supported.

Cisco TelePresence Conductor supports both ad hoc and meet-me voice and video conferencing. Cisco TelePresence Conductor dynamically selects the most appropriate Cisco TelePresence resource for each new conference. Ad hoc, “MeetMe” and scheduled voice and video conferences can dynamically grow and exceed the capacity of individual MCUs. One Cisco TelePresence Conductor appliance or Cisco TelePresence Conductor cluster has a system capacity of 30 MCUs or 2400 MCU ports. Up to three Cisco TelePresence Conductor appliances or virtualized applications may be clustered to provide greater resilience.

Cisco Meeting Server

The Cisco Meeting Server conference bridge solution allows Ad Hoc, Meet-Me, Conference Now, and Rendezvous conferences. This conference bridge offers premises-based audio, video, and web conferencing, and works with third-party on-premises infrastructure. It scales for small or large deployments. You can add capacity incrementally as needed, to ensure that you can support the current and future needs of your organization. This conference bridge provides advanced interoperability. Any number of participants can create and join meetings from:

- Cisco or third-party room or desktop video systems
- Cisco Jabber Client
- Cisco Meeting App (can be native or with a WebRTC compatible browser)
- Skype for Business

A minimum release of Cisco Meeting Server 2.0 is required to use the Cisco Meeting Server conference bridge.

The Cisco Meeting Server supports SIP as the signaling call control protocol. It has a built in Web Server that allows for complete configuration, control, and monitoring of the system and conferences. The Cisco Meeting Server provides XML management API over HTTP.

Note Cisco Meeting Server is enhanced to support AV1 codec and does not support H.265 video codec and Far End Camera Control (FECC).

Video Encryption

Unified Communications Manager supports encryption of audio, video, and other media streams so long as the individual endpoints involved in the communication also support encryption. Unified CM uses the Secure Real-Time Transport Protocol (SRTP) to encrypt the media streams. Some of the features include:

- Support for SIP and H.323 endpoints
- Support for encryption of main audio and video line while operating in Media Termination Point (MTP) passthru mode
- Support for multiple encryption methods
- Support for Session Description Protocol (SDP) crypto-suite session parameters in accordance with RFC 4568

To provide encrypted communications, encryption keys are exchanged between the endpoints and Unified Communications Manager during the SIP call setup. For this reason, the SIP signaling should be encrypted using TLS. During the initial call setup, the video endpoints exchange a list of encryption methods they support, select an encryption suite supported by both endpoints, and exchange encryption keys. If the endpoints cannot agree on a common encryption suite, the media streams are unencrypted and transported using the Real-Time Transport Protocol (RTP).

Note If the individual endpoints do not support encryption, the communication will take place using RTP.

Configure Interop with VCS

Perform the following steps on the SIP trunk that connects Unified Communications Manager to Cisco VCS to enable Unified CM to interoperate with a Cisco VCS.

Procedure

Step 1 From Cisco Unified CM Administration, choose **Device** > **Trunk**.

Step 2 Perform one of the following:

- Click **Find** and select an existing trunk.
- Click **Add New** to configure a new trunk.

Step 3 In the **Trunk Configuration** window, choose the **Trunk Type**, **Device Protocol**, **Trunk Service Type** that connects Unified Communications Manager to the Cisco VCS and click **Next**.

Step 4 In the **SIP Profile** drop-down list, choose **Standard SIP Profile for VCS**.

Step 5 In the **Normalization Script** drop-down list, choose **vcs-interop**.

Step 6 In the **Normalization Script** area, leave the **Parameter Name** and **Parameter Value** fields empty. If these fields are populated with values, delete the contents of the field.

Step 7 Click **Save**.

Video Features

The following video-related features are supported in SIP video networks:

- Binary Floor Control Protocol (BFCP)
- Encrypted iX Channel
- Far End Camera Control (FECC)

Endpoint Support for the Binary Floor Control Protocol

Unified Communications Manager provides support for the Binary Floor Control Protocol (BFCP) for specific Cisco and third-party video endpoints. BFCP allows users to share a presentation within an ongoing video conversation.

For more information, see chapter Configure Presentation Sharing using BFCP in Feature Configuration Guide for Cisco Unified Communications Manager.

Encrypted iX Channel

Unified Communications Manager supports an encrypted iX channel. The iX channel provides a reliable channel for multiplexing application media between SIP phones in a video conference. Encrypted iX Channel uses DTLS to add security to your deployment and ensures that the application media is sent over the iX Channel is private and cannot be viewed by intermediate parties who attempt to intercept media.

IOS MTP and RSVP agents in pass through mode also support encrypted iX Channel.

Configuration

To enable an encrypted iX Channel on Unified Communications Manager, you must:

- Check the **Allow iX Application Media** check box in the SIP Profile Configuration that is used by any intermediate SIP trunks. This setting turns on the iX channel negotiation.
- Configure the **Secure Call Icon Display Policy** service parameter to enable a secure lock icon. By default, the setting is **All media except BFCP and iX transports must be encrypted**.

Encryption Modes

There are two types of Session Description Protocol (SDP) offers thatUnified Communications Manager supports for iX Channel encryption for encrypted phones. This encryption type is driven by what the endpoints support and is not a configurable item in the Unified Communications Manager.

- **Best Effort Encryption**—The SDP offer is for an encrypted iX Channel, but falls back to a non-encrypted iX Channel if the SIP peers do not support it. This approach can be used if encryption is not mandatory in the solution.

  For example, encryption is mandatory within the cloud, and not in a single enterprise.

  Best-Effort iX Encryption

  m=application 12345 **UDP/UDT/IX** *

  a=setup:actpass

  a=fingerprint: SHA-1 <key>

- **Forced Encryption**—The SDP offer is for an encrypted iX Channel only. This offer is rejected if the SIP peers do not support iX Channel encryption. This approach can be used in deployments where encryption is mandatory between endpoints.

  For example, encryption is mandatory between the two SIP devices.

  Forced iX Encryption

  m=application 12345 **UDP/DTLS/UDT/IX** *

  a=setup:actpass

  a=fingerprint: SHA-1 <key>

By default, all Cisco IP Phones are set to offer Best Effort iX Encryption. However, you can reset this to Forced Encryption by setting the **Encryption Mode** to **On** within the Product-Specific Configuration of Cisco TelePresence endpoints, or by reconfiguring settings on the Cisco Meeting Server.

Non-Encrypted Modes

Unified Communications Managerenables negotiation of secure active control messages in media path from endpoints in a meeting when the endpoint may not be deployed in a fully secure mode. For example, if the endpoint is Off-Net and is registered with Unified CM in Mobile and Remote Access mode.

Prerequisite

Before you start using this feature, make sure that:

- System adheres to the export compliance requirement
- SIP trunk to the conference bridge is secure

Unified CM can negotiate the DTLS information in secure active control messages for non-secure endpoints or softphones and receive messages in the following ways:

- **Best Effort Encryption iX** to On-Premise registered endpoints or softphones
- **Forced iX Encryption** to Off-Premise registered endpoints or softphones

Far End Camera Control Protocol Support

The Far End Camera Control (FECC) protocol allows you to control a remote camera. Within video calls, FECC allows the party at one end of the call to control the camera at the far end. This control can include panning the camera from one side to the other, tilting the camera, or zooming in and out. For video conferences that use multiple cameras, FECC can be used to switch from one camera to another.

Unified Communications Manager supports the FECC protocol for video endpoints that are FECC-capable. Cisco Unified Communications Manager supports FECC for SIP-SIP calls or H.323-H.323 calls, but does not support FECC for SIP-H.323 calls. To support FECC, Unified Communications Manager sets the application media channel through SIP or H.323 signaling. After the media channel is established, the individual endpoints can communicate the FECC signaling.

QoS for Video Networks

Cisco Unified Communications Manager contains a number of administrative tools for managing Quality of Service (QoS) for video networks:

- Bandwidth Management—Mange bandwidth allocations for specific Regions and Locations
- Enhanced Locations Call Admission Control
- Session Level Bandwidth Modifiers
- Flexible DSCP Markings
- Alternate Routing

Bandwidth Management

Bandwidth allocations for audio and video calls are managed through regions and locations that you configure in Cisco Unified Communications Manager Administration.

The amount of bandwidth available for a specific call must be able to manage the combination of all media streams that are associated with the session, including voice, video, signaling, and any extra media, such as a BFCP presentation. Cisco Unified Communications Manager contains features able to manage bandwidth.

Enhanced Locations Call Admission Control

Enhanced Locations Call Admission Control (CAC) enables you to control the audio and video quality of calls over a wide-area (IP WAN) link by limiting the number of calls that are allowed on that link at the same time. For example, you can use call admission control to regulate the voice quality on a 56-kb/s frame relay line that connects your main campus and a remote site.

CAC verifies whether there is sufficient bandwidth available to complete a call. CAC can reject calls due to insufficient bandwidth.

In Unified Communications Manager, locations-based call admission control works in conjunction with regions to define the characteristics of a network link. Regions and locations work in the following manner:

- Regions allow the bandwidth of video calls to be set. The audio limit on the region may result in filtering out codecs with higher bit rates. However, for video calls, the video limit constrains the quality (resolution and transmission rate) of the video.
- Locations define the amount of total bandwidth available for all calls on that link. When a call is made on a link the regional value for that call must be subtracted from the total bandwidth allowed for that link.

For more information about call admission control, see chapter 'Configure Enhanced Locations Call Admission Control' in System Configuration Guide for Cisco Unified Communications Manager

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Session Level Bandwidth Modifiers

Unified Communications Manager provides location call admission control support for handling session level bandwidth modifiers. Session level bandwidth modifiers are communicated as part of the parameters in the SDP portion of the initial SIP signaling. These parameters indicate the maximum amount of bandwidth each endpoint will support for that type of call. These parameters are used, along with regions and locations settings, to set the bandwidth for each call.

During the initial call setup, both parties communicate to Unified Communications Manager their maximum allowed bandwidth for the call. Unified Communications Manager passes this communication to the other endpoint, but if the bandwidth that is specified by the endpoint is greater than the region setting, Unified Communications Manager replaces the value with the region bandwidth value.

Unified Communications Manager uses the following rules to determine the amount of bandwidth to allocate to a specific call:

- When Unified Communications Manager receives an Offer or Answer from an endpoint, it checks whether there is a session level bandwidth modifier in the SDP:

- If there is a session level bandwidth modifier, Unified Communications Manager retrieves the bandwidth value from the modifier. If there is more than one modifier type, it retrieves the modifier in the following order of preference: Transport Independent Application Specific (TIAS), Application Specific (AS), Conference Total (CT).
- If there is no session level bandwidth modifier, Unified Communications Manager retrieves the bandwidth value from the sum of the media level bandwidth modifiers.

- The allocated bandwidth is the maximum of what the two endpoints support up to the maximum value of the Region setting. The allocated bandwidth cannot exceed the region setting.

Unified Communications Manager uses the following logic when communicating with endpoints:

- When generating an Answer, Early Offer or Re-Invite Offer to an endpoint that contains more than one session level bandwidth modifier type (TIAS, AS, CT), Unified Communications Manager uses the same bandwidth value for each.
- When generating an answer, Unified Communications Manager uses the same session level bandwidth modifier type (TIAS, CT, AS) that was received in the initial offer
- For backward compatibility, the older Unified Communications Manager suppresses the Session Level Bandwidth Modifier when a video call is put on hold and music on hold (MOH) is inserted.

Video Resolution Support for SIP Phones

Cisco Unified Communications Manager supports the imageattr line in the SDP portion of the SIP header for higher resolution video calls. Cisco SIP phones that support w360p (640 x 360), such as the 9951, 9971, and 8961, automatically select the best resolution for video calls depending on the following criteria:

- If the session level bandwidth is greater than 800Kb/s and the imageattr[640 x 480] line in the SDP exists, then VGA is used.
- If the session level bandwidth is greater than 800Kb/s and the imageattr[640 x 480] line in the SDP does not exist, then w360p is used.
- If the session level bandwidth is less than 800Kb/s but greater than 480 bits per second and the imageattr[640 x 480] line exists, then VGA 15 frames per second is used.



Note If you currently have a Cisco IP Phone model 9951, 9971, or 8961 that supports w360p (640 x 360) video resolution and are upgrading to Cisco Unified Communications Manager release 8.5(1) or later, you may notice changes in the resolution of video calls. The w360p resolution was introduced at phone load 9.2(1).

The following video call flow is between two 9951 phones (Phone A and Phone B) without imageattr line support (for example, using Cisco Unified Communications Manager releases 8.0(1) and earlier):

1. Phone A sends a SIP message with an imageattr line in the SDP.
2. Cisco Unified Communications Manager deletes the imageattr line in the SDP and then sends the modified SIP message to Phone B.
3. Phone B attempts to send video with the w360p resolution because there is no imageattr line in the SDP portion of the SIP header.

The following video call flow is between two 9951 phones (Phone A and Phone B) with imageattr line support (for example, using Cisco Unified Communications Manager releases 8.5(1) and later):

1. Phone A sends a SIP message with the imageattr line in the SDP.
2. Cisco Unified Communications Manager does not delete the imageattr line and sends the SIP message to Phone B unchanged.
3. Phone B attempts to send video with the VGA resolution.

Alternate Routing

If an endpoint cannot obtain the bandwidth that it needs for a video call, a video call retries as an audio call for the default behavior. To use route/hunt lists or Automated Alternate Routing (AAR) groups to try different paths for such video calls, uncheck the Retry Video Call as Audio setting in the configuration settings for applicable gateways, trunks, and phones.

For more information, see Configure AAR Group section under Configure Call Routing chapter in the System Configuration Guide for Cisco Unified Communications Manager.

Flexible DSCP Markings

Differentiated Services Code Point (DSCP) packet marking, is used to specify the class of service for each packet. DSCP marking lets you prioritize certain types of calls or media over other types. For example, you can prioritize audio over video so that, even if you experience network bandwidth issues, audio calls should not experience bandwidth issues.

You can customize DSCP markings in either of these ways:

- Configure cluster wide service parameters to set the default DSCP settings for the cluster
- (Optional) For a subset of the DSCP categories, you can assign customized DSCP settings to devices via the SIP Profile. For the devices that use the profile, the customized settings override the service parameter defaults.

For more information on how to configure DSCP markings, see 'Configure Flexible DSCP Marking and Video Promotion' chapter in the Feature Configuration Guide for Cisco Unified Communications Manager.

Phone Configuration for Video Calls

The following setting for video-enabled devices affects video calls:

- Retry Video Call as Audio-By default, this check box remains checked. Thus, if an endpoint (phone, gateway, trunk) cannot obtain the bandwidth that it needs for a video call, call control retries the call as an audio call. This setting applies to the destination devices of video calls.
- Video Capabilities Enabled/disabled-This drop-down list box turns video capabilities on and off.

Conference Control for Video Conferencing

Unified Communications Manager supports the following conference controls capabilities:

- Roster/Attendee List
- Drop Participant
- Terminate Conference
- Show Conference Chairperson/Controller
- Continuous Presence

Unified Communications Manager also supports the following video conference capabilities for Skinny Client Control Protocol (SCCP) phones:

- Display controls for video conferences. The SCCP phones can choose to use the continuous presence or voice-activated mode to view the video conference. When a mode is chosen, a message gets sent to the bridge to indicate which mode to use on the video channel. Switching between modes does not require renegotiation of media.
- Display participant information such as the user name in the video stream. The system can use the participant information for other conferencing features such as roster.

For more information, see 'Encrypted iX Channel' chapter in the Security Guide for Cisco Unified Communications Manager.

Video Telephony and Cisco Unified Serviceability

Cisco Unified Serviceability tracks video calls and conferences by updating performance monitoring counters, video bridge counters, and call detail records (CDRs).

Performance Counters

Video telephony events cause updates to the following Cisco Unified Serviceability performance monitoring counters:

Cisco CallManager

- VCBConferenceActive
- VCBConferenceCompleted
- VCBConferenceTotal
- VCBOutOfConferences
- VCBOutOfResources
- VCBResourceActive
- VCBResourceAvailable
- VideoCallsActive
- VideoCallsCompleted
- VideoOutOfResources

Gatekeeper

- VideoOutOfResources

CiscoH.323

- VideoCallsActive
- VideoCallsCompleted

Cisco Locations

- RSVP VideoCallsFailed
- RSVP VideoReservationErrorCounts
- VideoBandwidthAvailable
- VideoBandwidthMaximum
- VideoOutOfResources

Cisco SIP

- VideoCallsActive
- VideoCallsCompleted

Cisco Vedio Conference Bridge

- ConferencesActive
- ConferencesAvailable
- ConferencesCompleted
- ConferencesTotal
- OutOfConferences
- OutOfResources
- ResourceActive
- ResourceAvailable
- ResourceTotal

Video Bridge Counters

Video conference events cause updates to these Cisco video conference bridge performance monitoring counters:

- ConferencesActive
- ConferencesAvailable

- ConferencesCompleted
- ConferencesTotal
- OutOfConferences
- OutOfResources
- ResourceActive
- ResourceAvailable
- ResourceTotal

These counters also display in the Cisco Unified Communications Manager object with the VCB prefix.

Call Detail Records (CDRs)

Video telephony events cause updates to Call Detail Records (CDRs) in Cisco Unified Serviceability. These CDRs include the following information:

- origVideoCap_Codec
- origVideoCap_Bandwidth
- origVideoCap_Resolution
- origVideoTransportAddress_IP
- origVideoTransportAddress_Port
- destVideoCap_Codec
- destVideoCap_Bandwidth
- destVideoCap_Resolution
- destVideoTransportAddress_IP
- destVideoTransportAddress_Port
- origRSVPStat
- destRSVPVideoStat
- origVideoCap_Codec_Channel2
- origVideoCap_Bandwidth_Channel2
- origVideoCap_Resolution_Channel2
- origVideoTransportAddress_IP_Channel2
- origVideoTransportAddress_Port_Channel2
- origVideoChannel_Role_Channel2
- destVideoCap_Codec_Channel2
- destVideoCap_Bandwidth_Channel2
- destVideoCap_Resolution_Channel2

- destVideoTransportAddress_IP_Channel2
- destVideoTransportAddress_Port_Channel2
- destVideoChannel_Role_Channel2

Call Management Records (CMRs)

Video telephony events cause updates to Call Management Records (CMRs) in Cisco Unified Serviceability. These CMRs include the following information:

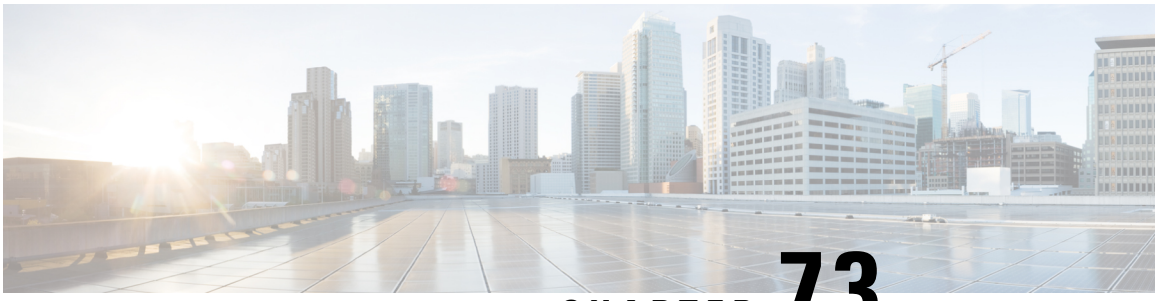
- videoContentType Text String
- videoDuration Integer
- numberVideoPacketsSent Integer
- numberVideoOctetsSent Integer
- numberVideoPacketsReceived Integer
- numberVideoOctetsReceived Integer
- numberVideoPacketsLost Integer
- videoAverageJitter Integer
- videoRoundTripTime
- videoOneWayDelay
- videoTransmissionMetrics



PART XVII

Emergency Call Routing Regulations

- The US Federal Communications Commission (FCC) Emergency Call Routing Regulations, on page 885



CHAPTER 73

The US Federal Communications Commission (FCC) Emergency Call Routing Regulations

- Emergency Call Routing Regulations Overview, on page 885
- Configure Emergency Call Routing Regulations, on page 887

Emergency Call Routing Regulations Overview

Emergency Call Routing Regulation provides us with information in compliance with the US FCC laws on how the Emergency Calls (911) are configured and routed in the US and non-US time zones.

The US FCC signed the following laws to ease public safety by encouraging and enabling the prompt deployment of a nationwide, seamless communications infrastructure for emergency services.

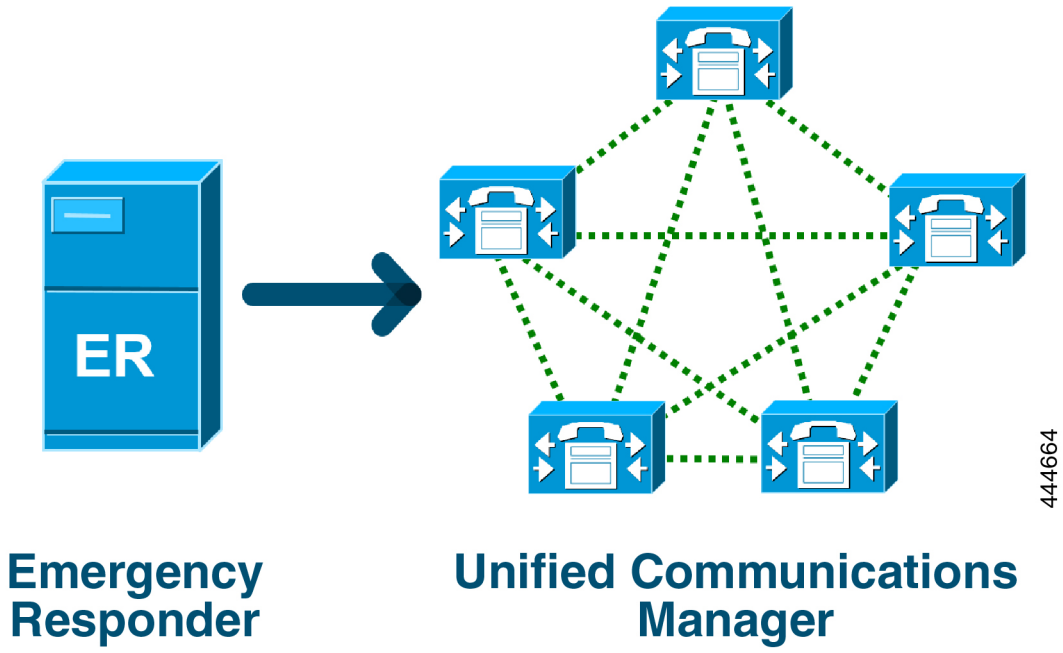
The US FCC signed the following laws on Emergency Call (911) routing:

- Kari's Law—This law applies to Multi-Line Telephone Systems (MLTS) that serve users in settings such as office buildings, campuses, and hotels. The FCC requires MLTS to enable users to dial 911 directly, without dialing a prefix reach an outside line and notifies the front desk or security office when emergency calls are made.
- Ray Baum's Act—Under section 506 of Ray Baum’s act, dispatch the location details (a street address, building number, floor number, room number) with emergency calls regardless of the technological platform used so that the 911 call centers receive the caller's location automatically and can dispatch responders more quickly.

For more information on FCC laws, see https://www.fcc.gov/mlts-911-requirements

An Emergency Responder effectively manages calls in the telephony network to respond and handle all emergency calls in compliance with local ordinances. It also dispatches location details and notifications are dispatched to the Unified Communications Manager.

The following image shows the connection between the Emergency Responder and Unified Communications Manager.



For more information on Cisco Emergency Responder, see Cisco Emergency Responder Administration Guide.

Unified Communications Manager as MLTS

The Cisco Unified Communications Manager Administration is an MLTS that has an inbuilt software to detect an absence of a direct 911 dial pattern for systems installed in the US time zone.

If the 911 route pattern isn't enabled, the Cisco Unified CM Administration home page displays an alert message: **You have not configured a direct dial 911 pattern on this system. Federal Communication Commission rules mandate that most multi-line telephone systems in the US have a direct dial 911 pattern.**



If the system installed in a non-US time zone, where FCC laws aren't applicable, Emergency Call Routing Regulations configuration page disabled in the Unified Communications Manager.

Note You should consult with their legal counselor on the applicability of FCC laws and acknowledge in the system.

Configure Emergency Call Routing Regulations

Emergency Call Routing Regulation configured on Unified Communications Manager to acknowledge and configure the direct dial 911 route pattern in compliance with the laws.

Before you begin

Ensure to take a backup after you accept and configure FCC Laws for future reference.

Procedure

Step 1 To access the **Emergency Call Routing Regulations** window, perform either of the following:

- From Cisco Unified CM Administration, choose**Advanced Features** > **Emergency Call Routing Regulations**
- Click the link available in the alert notification to configure the 911 route pattern on the home page.

Step 2 Check the **I have read the above notification,and I have consulted my legal counsel to determine my specific obligations** check box to acknowledge the notification.

Step 3 Check the **Take me to the 911 configuration page** check box and click **Submit** to set up direct 911 notification if the FCC laws are applicable. You are navigated to the **Route Pattern Configuration** window and by default, a 911 pattern is configured in the **Pattern Definition** section.

Step 4 Choose an appropriate gateway, route, or trunk from the **Gateway/Route List** drop-down list for the configured pattern. For more information on the other fields and their configurations, see Online Help.

Step 5 Click **Save**.

Note
If the system installed in the US time zone where FCC laws aren't applicable, acknowledge the law and check the **Disable any further notification regarding my 911 obligation** check box in the **Emergency Call Routing Regulations** window and click **Submit** to disable the 911 notifications.

If the laws aren't applicable, the administrator waives the notifications for future upgrades and new installations of the 911 route pattern.

The configured settings get preserved for future upgrade. The alert notification disappears on the home page and the **Emergency Call Routing Regulations** window gets disabled.

If the system has already created a 911 route pattern during upgrade or the time zone changed from non-US to US timezone then the acknowledge page is grayed out.