Features, Templates, Services, and User Setup

Features, Templates, Services, and User Setup Overview

After you install conferences stations in your network, configure network settings, and add each Cisco Unified IP Conference Phone to Cisco Unified Communications Manager, you must use the Cisco Unified Communications Manager Administration application to configure telephony features, optionally modify conference phone templates, set up services, and assign users.

This chapter provides an overview of these configuration and setup procedures. Cisco Unified Communications Manager documentation provides detailed instructions for these procedures.

For suggestions about how to provide users with information about features, and what information to provide, see Internal Support Website.

For information about setting up conference phones in non-English environments, see International User Support.

Available Telephony Features

After you add Cisco Unified IP Conference Phones to Cisco Unified Communications Manager, you can add functionality. The following table includes a list of supported telephony features. You can use Cisco Unified Communications Manager Administration to configure many of these features. The Reference column lists Cisco Unified Communications Manager and other documentation that contains configuration procedures and related information.

For information about the use of these features on the phone, see the *Cisco Unified IP Conference Phone 8831 and 8831NR User Guide*.

Note Cisco Unified Communications Manager Administration also provides several service parameters that are used to configure various telephony functions. For more information about access and configuration of service parameters, see the *Cisco Unified Communications Manager Administration Guide*.

For more information on the functions of a service, select the name of the parameter or the question mark help button in the Service Parameter Configuration window in CUCM Administration.

Table 1: Telephony Features for Cisco Unified IP Conference Phone 8831

Feature	Description	Configuration Reference
Agent Greeting	Allows an agent to create and update a prerecorded greeting that plays at the beginning of a call, such as a customer call, before the agent begins the conversation with the caller. The agent can prerecord a single greeting or multiple greetings as needed. To enable Agent Greeting in the Cisco Unified Communications Manager Administration application, choose **Device** > **Phone**, locate the IP Phone that you want to configure. Scroll to the Device Information Layout pane and set Built In Bridge to **On** or **Default**. If Built In Bridge is set to Default, in the Cisco Unified Communications Manager Administration application, choose **System** > **Service Parameter** and select the appropriate Server and Service. Scroll to the Clusterwide Parameters (Device - Phone) pane and set **Builtin Bridge Enable** to **On**.	For more information, see the following: • *Features and Services Guide for Cisco Unified Communications Manager*, "Barge and Privacy" • *Cisco Unified Communications Manager System Guide*, "Cisco Unified IP Phones"
Any Call Pickup	Allows users to pick up a call on any line in their call pickup group, regardless of how the call was routed to the phone.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, "Call Pickup" chapter.

Feature	Description	Configuration Reference
Audible Message Waiting Indicator (AMWI)	A stutter tone indicates that a user has one or more new voice messages on a line. **Note** The stutter tone is line-specific. You hear it only when using the line with the waiting messages.	For more information, see *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” chapter.
Auto Answer	Connects incoming calls automatically after a ring or two.	For more information, see *Cisco Unified Communications Manager Administration Guide*, “Directory Number Setup” chapter.
Auto Pickup	Allows a user to use one-touch pickup functionality for call pickup features.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Call Pickup” chapter.
Block External to External Transfer	Prevents users from transferring an external call to another external number.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “External Call Transfer Restrictions” chapter.
Call Back	Provides users with an audio and visual alert on the phone when a busy or unavailable party becomes available.	For more information, see the following: • *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” • *Features and Services Guide for Cisco Unified Communications Manager*, “Cisco Call Back”
Call Display Restrictions	Determines the information that will display for calling or connected lines, depending on the parties who are involved in the call.	For more information, see: • *Cisco Unified Communications Manager Administration Guide*, “Cisco Unified IP Phone Setup” • *Cisco Unified Communications Manager System Guide*, “Understanding Route Plans” • *Features and Services Guide for Cisco Unified Communications Manager*, “Call Display Restrictions”

Feature	Description	Configuration Reference
Call Forward	Allows users to redirect incoming calls to another number. Call Forward options include Call Forward All, Call Forward Busy, Call Forward No Answer, and Call Forward No Coverage.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, “Directory Number Setup” • *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” • Customize Cisco Unified Communications Manager Self Care Portal display
Call Forward All Loop Breakout	Detects and prevents Call Forward All loops. When a Call Forward All loop is detected, the Call Forward All configuration is ignored and the call rings through.	For more information, see *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” chapter.
Call Forward All Loop Prevention	Prevents a user from configuring a Call Forward All destination directly on the phone that creates a Call Forward All loop or that creates a Call Forward All chain with more hops than the existing *Forward Maximum Hop* Count service parameter allows.	For more information, see *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” chapter.
Call Forward Configurable Display	Allows you to specify information that appears on a phone when a call is forwarded. This information can include the caller name, caller number, redirected number, and original dialed number.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, “Directory Number Setup” • *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones”
Call Forward Destination Override	Allows you to override Call Forward All (CFA) in cases where the CFA target places a call to the CFA initiator. This feature allows the CFA target to reach the CFA initiator for important calls. The override works whether the CFA target phone number is internal or external.	For more information, see the *Cisco Unified Communications Manager System Guide*, “Directory Numbers” chapter.
Call Forward Notification	Allows you to configure the information that the user sees when receiving a forwarded call.	For more information, see Call Forward Notification setup.
Call History for Shared Line	Allows you to view shared line activity in the phone Call History. This feature will: • Log missed calls for a shared line • Log answered calls for a shared line	For more information, see Enable Call History for shared line.

Feature	Description	Configuration Reference
Call Park	Allows users to park (temporarily store) a call and then retrieve the call by using another phone in the Cisco Unified Communications Manager system.	For more information, see the *Features and Services Guide for Cisco Unified Communications Manager*, “Call Park and Directed Call Park” chapter.
Call Pickup	Allows users to redirect a call that is ringing on another phone within their pickup group to their phone. You can configure an audio and visual alert for the primary line on the phone. This alert notifies the users that a call is ringing in their pickup group.	For more information, see the *Features and Services Guide for Cisco Unified Communications Manager*, “Call Pickup” chapter.
Call Recording	Allows a supervisor to record an active call. The user might hear a recording audible alert tone during a call when it is being recorded. When a call is secured, the security status of the call is displayed as a lock icon on Cisco Unified IP Phones. The connected parties might also hear an audible alert tone that indicates the call is secured and is being recorded. **Note** When an active call is being monitored or recorded, you can receive or place intercom calls; however, if you place an intercom call, the active call will be put on hold, which causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the party whose call is being monitored must resume the call.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Monitoring and Recording” chapter.
Call Waiting	Indicates (and allows users to answer) an incoming call that rings while on another call. Incoming call information appears on the phone display.	For more information, see: • *Cisco Unified Communications Manager System Guide*, “Directory Numbers” • Phone Call Waiting setup

Feature	Description	Configuration Reference
Caller ID	Caller identification such as a phone number, name, or other descriptive text appear on the phone display.	For more information, see: • *Cisco Unified Communications Manager Administration Guide*, "Cisco Unified IP Phone Setup" • *Cisco Unified Communications Manager System Guide*, "Understanding Route Plans" • *Features and Services Guide for Cisco Unified Communications Manager*, "Call Display Restrictions" • *Cisco Unified Communications Manager Administration Guide*, "Directory Number Setup"
Caller ID Blocking	Allows a user to block their phone number or email address from phones that have caller identification enabled.	For more information, see: • *Cisco Unified Communications Manager System Guide*, "Understanding Route Plans" • *Cisco Unified Communications Manager Administration Guide*, "Directory Number Setup"
Calling Party Normalization	Calling party normalization presents phone calls to the user with a dialable phone number. Any escape codes are added to the number so that the user can easily connect to the caller again. The dialable number is saved in the call history and can be saved in the Personal Address Book.	For more information, see Calling Party Normalization.
cBarge	Allows a user to join a nonprivate call on a shared phone line. cBarge adds a user to a call and converts it into a conference, allowing the user and other parties to access conference features	For more information, see: • *Cisco Unified Communications Manager Administration Guide*, "Cisco Unified IP Phone Setup" • *Cisco Unified Communications Manager System Guide*, "Cisco Unified IP Phones" • *Features and Services Guide for Cisco Unified Communications Manager*, "Barge and Privacy"

Feature	Description	Configuration Reference
Cisco Extension Mobility	Allows users to temporarily access their Cisco Unified IP Phone configuration such as line appearances and services from a shared Cisco Unified IP Phone by logging into the Cisco Extension Mobility service on that phone when they log into the Cisco Extension Mobility service on that phone. Cisco Extension Mobility can be useful if users work from a variety of locations within your company or if they share a workspace with coworkers.	For more information, see “ Extension Mobility” chapter in the *Cisco Unified Communications Manager Features and Services Guide*.
Cisco Unified Communications Manager Express (Unified CME) Version Negotiation	The Cisco Unified Communication Manager Express uses a special tag in the information sent to the phone to identify itself. This tag enables the phone to provide services to the user that the switch supports.	For more information, see: • Cisco Unified Communications Manager Express System Administrator Guide
Cisco WebDialer	Allows users to make calls from web and desktop applications.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Web Dialer” chapter.
Conference	• Allows a user to talk simultaneously with multiple parties by calling each participant individually. Conference features include Conference and Meet Me. • Allows a noninitiator in a standard (ad hoc) conference to add or remove participants; also allows any conference participant to join together two standard conferences on the same line.	The service parameter, Advance Adhoc Conference, (disabled by default in Cisco Unified Communications Manager Administration) allows you to enable these features. For information on conferences, see *Cisco Unified Communications Manager System Guide*, “Conference Bridges” chapter. For more information, see *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” chapter. **Note** Be sure to inform your users whether these features are activated.
CTI Applications	A computer telephony integration (CTI) route point can designate a virtual device to receive multiple, simultaneous calls for application-controlled redirection.	For more information, see *Cisco Unified Communications Manager Administration Guide*, “CTI Route Point Setup” chapter.

Feature	Description	Configuration Reference
Device Invoked Recording	Provides end users with the ability to record their telephone calls via a softkey. In addition administrators may continue to record telephone calls via the CTI User Interface.	For more information, see Enable Device Invoked Recording.
Directed Call Park	Allows a user to transfer an active call to an available directed call park number that the user dials. **Note** If you implement Directed Call Park, avoid configuring the Park softkey. This prevents users from confusing the two Call Park features.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Call Park and Directed Call Park” chapter.
Directed Call Pickup	Allows a user to answer a call that is ringing on a particular directory number.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Call Pickup” chapter.
Direct Transfer	Allows users to connect two calls to each other without remaining on the line.	For more information, see *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phones” chapter.
Distinctive Ring	Users can customize how their phone indicates an incoming call and a new voice mail message.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Call Pickup” chapter.
Divert	Allows a user to transfer a ringing, connected, or held call directly to a voice-messaging system. When a call is diverted, the line becomes available to make or receive new calls.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Immediate Divert” chapter.

Feature	Description	Configuration Reference
Do Not Disturb (DND)	When DND is turned on, either no audible rings occur during the ringing-in state of a call, or no audible or visual notifications of any type occur. The following DND-related parameters are configurable in Cisco Unified Communications Manager Administration: • Do Not Disturb: This check box allows you to enable DND on a per-phone basis. Use **Cisco Unified Communications Manager Administration** > **Device** > **Phone** > **Phone Configuration**. • DND Incoming Call Alert: Choose the type of alert to play, if any, on a phone for incoming calls when DND is active. This parameter is located on both the Common Phone Profile page and the Phone configuration page (Phone Configuration window value takes precedence).	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Do Not Disturb” chapter.
EnergyWise	Enables an IP Phone to sleep (power down) and wake (power up) at predetermined times, to promote energy savings.	For more information, see EnergyWise on the Cisco Unified IP Phone setup.
Enhanced Room Coverage	Optional microphone extension kits provide enhanced room coverage that can be further expanded by linking two units together in Linked Mode.	For more information see: • *Cisco Unified IP Conference Phone 8831 User Guide for Cisco Unified Communication Manager*, “Conference Phone Link Mode” chapter. • *Cisco Unified IP Conference Phone 8831 User Guide for Cisco Unified Communication Manager*, “Enhanced Room Coverage” chapter.
Fast Dial Service	Allows a user to enter a Fast Dial code to place a call. Fast Dial codes can be assigned to phone numbers or Personal Address Book entries. (See “Services” in this table.)	For more information, see Modify phone button template for PAB or Fast Dial.
Group Call Pickup	Allows a user to answer a call that is ringing on a directory number in another group.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Call Pickup” chapter.

Feature	Description	Configuration Reference
Hold Reversion	Limits the amount of time that a call can be on hold before reverting back to the phone that put the call on hold and alerting the user. Reverting calls are distinguished from incoming calls by a single ring (or beep, depending on the new call indicator setting for the line). This notification repeats at intervals if not resumed. A call that triggers Hold Reversion also displays an animated icon in the call bubble. You can configure call focus priority to favor incoming or reverting calls.	For more information about configuring this feature, see *Features and Services Guide for Cisco Unified Communications Manager*, "Hold Reversion" chapter.
Hold Status	Enables phones with a shared line to distinguish between the local and remote lines that placed a call on hold.	No configuration is required.
Hold/Resume	Allows the user to move a connected call from an active state to a held state.	• Requires no configuration, unless you want to use Music On Hold. See "Music On Hold" in this table for information. • See "Hold Reversion" in this table.
HTTP Download	Enhances the file download process to the phone to use HTTP by default. If the HTTP download fails, the phone reverts to using the TFTP download.	No configuration is required.
Jitter Buffer	The Jitter Buffer feature handles jitter from 10 milliseconds (ms) to 1000 ms for both audio and video streams.	No configuration required.
Linked Mode	Enhances the audio coverage area by using a daisy cable to connect two conference phone Sound Base units. When linked, voice, dial tone, ringer and base LED features are synchronized between the two devices.	For more information see: • *Cisco Unified IP Conference Phone 8831 User Guide*, "Conference Phone Link Mode"
Malicious Caller Identification (MCID)	Allows users to notify the system administrator about suspicious calls that are received.	For more information, see: • *Cisco Unified Communications Manager System Guide*, "Cisco Unified IP Phones" • *Features and Services Guide for Cisco Unified Communications Manager*, "Malicious Call Identification"
Meet Me Conference	Allows a user to host a Meet Me conference in which other participants call a predetermined number at a scheduled time.	For more information, see *Cisco Unified Communications Manager Administration Guide*, "Meet-Me Number and Pattern Setup" chapter.

Feature	Description	Configuration Reference
Message Waiting	Defines directory numbers for message waiting on and off indicators. A directly-connected voice-message system uses the specified directory number to set or to clear a message waiting indication for a particular Cisco Unified IP Phone.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, “Message Waiting Setup” • *Cisco Unified Communications Manager System Guide*, “Voice Mail Connectivity to Cisco Unified Communications Manager”
Message Waiting Indicator	Displays a New Voicemail status message on the phone screen.	For more information see: • *Cisco Unified Communications Manager Administration Guide*, “Message Waiting Setup” • *Cisco Unified Communications Manager System Guide*, “Voice Mail Connectivity to Cisco Unified Communications Manager”
Missed Call Logging	Allows a user to specify whether missed calls will be logged in the missed calls directory for a given line appearance.	For more information, see *Cisco Unified Communications Manager Administration Guide*, “Directory Number Setup” chapter.
Mobile Connect	Enables users to manage business calls using a single phone number and pick up in-progress calls on the desk phone and a remote device such as a mobile phone. Users can restrict the group of callers according to phone number and time of day.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Cisco Unified Mobility” chapter.
Mobile Voice Access	Extends Mobile Connect capabilities by allowing users to access an interactive voice response (IVR) system to originate a call from a remote device such as a cellular phone.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Cisco Unified Mobility” chapter.

Feature	Description	Configuration Reference
Monitoring and Recording	Allows a supervisor to silently monitor an active call. The supervisor cannot be heard by either party on the call. The user might hear a monitoring audible alert tone during a call when it is being monitored. When a call is secured, the security status of the call is displayed as a lock icon on Cisco Unified IP Phones. The connected parties might also hear an audible alert tone that indicates the call is secured and is being monitored. **Note** When an active call is being monitored or recorded, the use can receive or place intercom calls; however, if the user place an intercom call, the active call will be put on hold, which causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the party whose call is being monitored must resume the call.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Monitoring and Recording” chapter.
Mute	Mutes the conference phone from the Sound Base, DCU or connected external microphones.	Requires no configuration.
No Alert Name	Makes it easier for end users to identify transferred calls by displaying the original caller’s phone number. The call appears as an Alert Call followed by the caller’s telephone number.	Requires no configuration.
Onhook Dialing	Allows a user to dial a number without going off hook.	For more information, see *Cisco Unified IP Conference Phone 8831 User Guide*.
Other Group Pickup	Allows a user to answer a call ringing on a phone in another group that is associated with the user's group.	For more information, see *Features and Services Guide for Cisco Unified Communications Manager*, “Call Pickup” chapter.
Phone Display Message for Extension Mobility Users	This feature enhances the phone interface for the Extension Mobility user by providing friendly messages.	No configuration required.
Plus Dialing	Allows the user to dial E.164 numbers prefixed with a plus (+) sign. To dial the + sign, the user needs to press and hold the star (*) key for at least 1 second. This applies to dialing the first digit for an on-hook (including edit mode) or off-hook call.	No configuration required.

Feature	Description	Configuration Reference
Privacy	Prevents users who share a line from adding themselves to a call and from viewing information on their phone display about the call of the other user.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, "Cisco Unified IP Phone Setup" • *Cisco Unified Communications Manager System Guide*, "Cisco Unified IP Phones" • *Features and Services Guide for Cisco Unified Communications Manager*, "Barge and Privacy"
Quality Reporting Tool (QRT)	Allows users to submit information about problem phone calls by pressing a button. QRT can be configured for either of two user modes, depending upon the amount of user interaction desired with QRT.	For more information, see: • *Cisco Unified Communications Manager System Guide*, "Cisco Unified IP Phones" • *Features and Services Guide for Cisco Unified Communications Manager*, "Quality Report Tool"
Recording Tone	Indicates if a recording tone is enabled for the phone.	For more information, see Recording Tone, on page 20
Redial	Allows users to call the most recently dialed phone number by pressing a button or the Redial softkey.	Requires no configuration.
Reroute Direct Calls to Remote Destination to Enterprise Number	Reroutes a direct call to a user's mobile phone to the enterprise number (desk phone). For an incoming call to remote destination (mobile phone), only remote destination rings; desk phone does not ring. When the call is answered on their mobile phone, the desk phone displays a Remote In Use message. During these calls, users can make use of various features of their mobile phone.	For more information, see the *Features and Services Guide for Cisco Unified Communications Manager*, "Cisco Unified Mobility" chapter.

Feature	Description	Configuration Reference
Remote Port Configuration	Allows the administrator to configure the speed and duplex function of the phone Ethernet ports remotely by using Cisco Unified Communications Manager Administration. This enhances the performance for large deployments with specific port settings. **Note** If the ports are configured for Remote Port Configuration in Cisco Unified Communications Manager, the data cannot be changed on the phone.	To configure the parameter in the Cisco Unified Communications Manager Administration application, choose **Device** > **Phone**, select the appropriate IP phone, and scroll to the Product Specific Configuration Layout pane (Switch Port Remote Configuration or PC Port Remote Configuration). To configure the setting on multiple phones simultaneously, configure the remote configuration in either Enterprise Phone Configuration (**System** > **Enterprise Phone Configuration**) or Common Phone Profile Configuration (**Device** > **Device Settings** > **Common Phone Profile**. (Switch Port Remote Configuration or PC Port Remote Configuration.)
Ringtone Setting	Identifies ring type used for a line when a phone has another active call.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, “Directory Number Setup” • *Cisco Unified Communications Manager Administration Guide*, “Custom Phone Rings”
Secure Conference	• Allows secure phones to place conference calls using a secured conference bridge. • As new participants are added by using Conf, Join, cBarge, Barge softkeys or MeetMe conferencing, the secure call icon displays as long as all participants use secure phones. • The Conference List displays the security level of each conference participant. Initiators can remove nonsecure participants from the Conference List. (Non-initiators can add or remove conference participants if the Advanced Adhoc Conference Enabled parameter is set.)	For more information about security, see Supported Security Features. For additional information, see the following: • *Cisco Unified Communications Manager System Guide*, “Conference Bridges” • *Cisco Unified Communications Manager Administration Guide*, “Conference Bridge Setup” • *Cisco Unified Communications Manager Security Guide*
Services	Allows you to use the Cisco Unified IP Phone Services Configuration menu in Cisco Unified Communications Manager Administration to define and maintain the list of phone services to which users can subscribe.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide*, “Cisco Unified IP Phone Setup” • *Cisco Unified Communications Manager System Guide*, “Cisco Unified IP Phone Services”

Feature	Description	Configuration Reference
Shared Line	Allows a user to have multiple phones that share the same phone number or allows a user to share a phone number with a coworker.	For more information, see *Cisco Unified Communications Manager System Guide*, “Directory Numbers” chapter.
Special Sequence for Factory Reset	Allows the phone manufacturer to reset the factory parameters.	No configuration required.
Time-of-Day Routing	Restricts access to specified telephony features by time period.	For more information, see: • *Cisco Unified Communications Manager Administration Guide*, “Time Period Setup” • *Cisco Unified Communications Manager System Guide*, “Time-of-Day Routing”
Time Zone Update	Updates the Cisco Unified IP Phone with time zone changes.	For more information, see *Cisco Unified Communications Manager Administration Guide*, “Date and Time Group Setup” chapter.
TLS Enhancements	Added in Firmware Release 10.3(1)SR3. Improved security with the parameter **Disable TLS1.0 and TLS1.1 for web access** in Cisco Unified Communications Manager. When set to enabled, the phones support only TLS1.2 mode. When set to disabled, TLS1.0, TLS1.1 and TLS1.2 are supported. This field applies to any phone or group of phones that function as a HTTPs server.	
Transfer	Allows users to redirect connected calls from their phones to another number.	Some JTAPI/TAPI applications are not compatible with the Join and Direct Transfer feature implementation on the conference phone and you may need to configure the Join and Direct Transfer Policy to disable join and direct transfer on the same line or possibly across lines.
Transfer - Direct Transfer	Transfer: The first invocation of Transfer will always initiate a new call by using the same directory number, after putting the active call on hold. Direct Transfer: This transfer joins two established calls (call is in hold or in connected state) into one call and drops the feature initiator from the call. Direct Transfer does not initiate a consultation call and does not put the active call on hold.	Some JTAPI/TAPI applications are not compatible with the Join and Direct Transfer feature implementation on the conference phone and you may need to configure the Join and Direct Transfer Policy to disable join and direct transfer on the same line or possibly across lines. For more information, see the <xref Join and Direct Transfer Policy>.

Feature	Description	Configuration Reference
UCR 2008	The conference phone supports Unified Capabilities Requirements (UCR) 2008 by providing support for 80-bit SRTCP Tagging. As an administrator, you must set up specific parameters in Cisco Unified Communications Manager Administration.	See UCR 2008 setup.
Voice Message System	Enables callers to leave messages if calls are unanswered.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, “Cisco Voice-Mail Port Setup” • *Cisco Unified Communications Manager System Guide*, “Voice Mail Connectivity to Cisco Unified Communications Manager”
Wideband Ringtone	The Wideband Ringtone feature supports 10 ringtones: 4 embedded in the phone firmware and 6 downloaded from the Cisco Unified Communications Manager.	For more information, see the following: • *Cisco Unified Communications Manager Administration Guide*, “Directory Number Setup” • *Features and Services Guide for Cisco Unified Communications Manager*, “Custom Phone Rings”
Wireless Microphone Frequency Lock	Provides a secure DECT frequency for wireless microphones by locking down the Wireless Region setting.	No configuration required for Firmware release 10.3(1) and later. Earlier releases must upgrade to the current release to have the region setting locked. **Note** Once you have configured the Wireless Region setting, it cannot be updated. Further configuration of this setting requires an RMA.

Corporate and Personal Directory Setup

The **Contacts** softkey on the conference phone gives users access to directory features. These directories can include:

- Corporate Directory: Supports a global corporate directory that users can access on the conference phone to lookup phone numbers of coworkers. You must enable and configure the corporate directories before they can be used.

- Personal Directory: Supports a personal address book (PAB) that allows a user to store a set of personal numbers. You must enable the feature and provide the user with a PIN and User ID to configure the features.

Corporate Directory Setup

Cisco Unified Communications Manager uses a Lightweight Directory Access Protocol (LDAP) directory to store authentication and authorization information about users of Cisco Unified Communications Manager applications that interface with Cisco Unified Communications Manager. Authentication establishes a user's right to access the system. Authorization identifies the telephony resources that a user is permitted to use, such as a specific teleconference phone extension.

To install and set up these features, see *Installing and Configuring the Cisco Customer Directory Configuration Plugin*. This document guides you through the configuration process for integrating Cisco Unified Communications Manager with Microsoft Active Directory and Netscape Directory Server. For more information, see "Understanding Directory Numbers" in the *Cisco Unified Communications Manager System Guide*.

After the LDAP directory configuration is complete, users can access the Corporate Directory service on the conference phone to find co-workers in the corporate directory.

Personal Directory Setup

Personal Directory consists of the following features:

- Personal Address Book (PAB)
- Personal Fast Dials (Fast Dials)

Users can access Personal Directory features by these methods:

- From a web browser: Users can access the PAB from Cisco Unified Communications Manager Self Care Portal.
- From the conference phone: Users can choose **Contacts** > **Personal Directory** to access the PAB and Fast Dials features from their conference phones.

To configure Personal Directory from a web browser, users must access Cisco Unified Communications Manager Self Care Portal. You must provide users with a URL and login information.

Button Templates

Using Cisco Unified Communications Manager Administration, you can assign button templates to conference phones. Cisco Unified Communications Manager contains the Standard 8831 button template for the Cisco Unified IP Conference Phone 8831 and 8831NR. When you assign the Standard 8831 button template to a conference phone, no buttons are added, but the **Privacy** softkey can be enabled from this template.

If the button feature is not configurable using the Button Template it may be a softkey feature that is configurable using a softkey template. For more information on softkey templates, see Softkey Templates, on page 18.

For more information on button templates, see the *Cisco Unified Communications Manager Administration Guide* and *Cisco Unified Communications Manager System Guide*.

Add Phone Button Template

To add a button template,

Procedure

Step 1 Log on to Cisco Unified Communications Manager Administration.

Step 2 Select **Device** > **Device Settings** > **Phone Button Template**.

Step 3 Select **Add New**.

Step 4 To assign a button template to a conference phone, choose **Device** > **Phone** to select the conference phone.

Step 5 In the Phone Configuration window, select a button template from the Phone Button Template drop-down list.

Softkey Templates

Using Cisco Unified Communications Manager Administration, you can manage softkeys associated with applications that are supported by the conference phone. Cisco Unified Communications Manager supports the Standard User and Standard Feature softkey templates.

An application that supports softkeys can have one or more standard softkey templates associated with it. You can modify a standard softkey template by making a copy of it, giving it a new name, and making updates to that copied softkey template. You can also modify a nonstandard softkey template.

To configure softkey templates, select **Device** > **Device Settings** > **Softkey Template** from Cisco Unified Communications Manager Administration. To assign a softkey template to a conference phone, use the Softkey Template field in the Cisco Unified Communications Manager Administration Phone Configuration page.

The Cisco Unified IP Conference Phone does not support all the softkeys that are configurable in Softkey Template Configuration on Cisco Unified Communications Manager Administration. The following table lists the features and softkeys that can be configured on a softkey template, and note whether it is supported on the conference phone.

Note Cisco Unified Communications Manager allows you to configure any softkey in a softkey template, but unsupported softkeys do not display on the phone.

Table 2: Configurable Softkeys

Feature	Configurable Softkeys in the Softkey Template Configuration	Supported as a Softkey	Notes
Answer	Answer (Answer)	Yes	—

Feature	Configurable Softkeys in the Softkey Template Configuration	Supported as a Softkey	Notes
Barge	Barge (Barge)	Yes	Configure as a softkey.
Call Back	Call Back (CallBack)	Yes	—
Call Forward All	Forward All (cfwdAll)	Yes	Phone displays **Fwd ALL** or **Fwd Off**.
Call Park	Call Park (Park)	Yes	—
Call Pickup	Pick Up (Pickup)	Yes	Configure as a softkey.
cBarge	Conference Barge (cBarge)	Yes	Configure as a softkey.
Conference	Conference (Conf)	Yes	Phone displays **Conference** when a call is connected.
Conference List	Conference List (ConfList)	Yes	Phone displays **ConfList** when in a conference.
iDivert	Immediate Divert (iDivert)	Yes	Phone displays **Divert**.
Do Not Disturb	Toggle Do Not Disturb (DND)	Yes	Configure as a softkey.
End Call	End Call (EndCall)	Yes	Phone displays **Cancel** if the call is not answered.
Group Pickup	Group Pick Up (GPickUp)	Yes	Configure as a softkey.
Hold	Hold (Hold)	Yes	
Join	Join (Join)	No	—
Malicious Call Identification	Toggle Malicious Call Identification (MCID)	Yes	Configure Malicious Call Identification as a softkey.
Meet Me	Meet Me (MeetMe)	Yes	Configure as a softkey.
Mobile Connect	Mobility (Mobility)	Yes	Configure Mobile Connect as a softkey.
New Call	New Call (NewCall)	Yes	Phone displays **New Call**.
Other Pickup	Other Pickup (oPickup)	Yes	Configure as a softkey.

Feature	Configurable Softkeys in the Softkey Template Configuration	Supported as a Softkey	Notes
Quality Reporting Tool	Quality Reporting Tool (QRT)	Yes	Configure Quality Reporting Tool as a softkey.
Redial	Redial (Redial)	Yes	Phone displays **Redial** when Off Hook.
Remove Last Conference Participant	Remove Last Conference Participant (Remove)	Yes	Phone displays **Remove** when a participant is selected.
Resume	Resume (Resume)	Yes	—
Transfer	Transfer	Yes	Phone displays **Transfer** when a call is connected.

For more information, see *Cisco Unified Communications Manager Administration Guide*, “Softkey Template Configuration” and the *Cisco Unified Communications Manager System Guide*, “Softkey Template”.

Set Up Softkey Templates

Procedure

Step 1 Log on to Cisco Unified Communications Manager Administration.

Step 2 Select **Device** > **Device Settings** > **Softkey Template**.

Step 3 To assign a softkey template to a conference station, choose **Device** > **Phone** to select the conference station.

Step 4 In the Phone Configuration window, select a softkey template from the Softkey Template drop-down list.

Recording Tone

The Recording Tone feature indicates whether a recording tone is enabled or disabled for the phone. If this feature is enabled, all parties on a call hear the tone being played, regardless of whether the call is recorded or not. The tone first sounds when a call is answered.

Note Other related parameters — the recording tone frequency (in hz), the duration of the beep tone, and the beep tone interval (how often the beep tone plays) — are defined on a per-Network Locale basis in the xml file that defines tones. This xml file is named tones.xml or g3-tones.xml.

Table 3: Recording Tone Parameters

Parameter	Note
Recording Tone Local Volume	Indicates the volume for the recording tone that the local party receives, if the local party phone has the Recording Tone option enabled. This setting applies for each listening device, including the handset, speakerphone, and headset. Range: 0 percent (no tone) to 100 percent (same level as current volume setting on the phone).
Recording Tone Remote Volume	Indicates the volume for the recording tone that the remote party receives. The remote party is the party who is on a call with the party whose phone has the Recording Tone option enabled. Range: 0 percent to 100 percent. (0 percent is -66 dBM and 100 percent is -3 dBM). Default: 50 percent.
Recording Tone Duration	Indicates the length of time in milliseconds that the tone plays. If the value is less than one third of the interval, this value overrides the default provided that the Network Locale provides. Range: 0 to 3000 **Note** For some Network Locales that use a complex cadence, this setting applies only to the first recording tone.

Enable Recording Tone

Before you begin

You must be familiar with the Recording Tone parameters before you enable this feature.

Note You may want to notify your users if you enable this option. The default setting is Disabled.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, go to **Device** > **Phone**.
Step 2 Select **Enable** from the pulldown menu In the Recording Tone section.
Step 3 Enter a value between 0 and 100 per cent in the Recording Tone Local Volume field.
Step 4 Enter a value between 0 and 100 per cent in the Recording Tone Remote Volume field.
Step 5 Click **Save**.

Enable Device Invoked Recording

Configure the Device Invoked Recording feature from Cisco Unified Communications Manager. To enable this feature, perform the following steps.

Procedure

Step 1 Set the Built In Bridge to **On**.

Step 2 Set Privacy to **Off**.

Step 3 Set Recording Option to **Selective Call Recording Enabled**.

Step 4 Select the appropriate Recording Profile.

Enable Call History for Shared Line

For more information, see *Cisco Unified Communications Manager Administration Guide*.

Note For the purposes of this procedure, all references to a phone, device, or IP Phone apply to the conference phone.

Procedure

Step 1 Go to Cisco Unified CM Administration and choose **Device** > **Phone**.

Step 2 Find your conference phone in the list of phones associated with the Cisco Unified CM.

Step 3 Click on the Device Name of the phone.

The Phone Configuration window appears.

Step 4 Go to Product Specific Configuration Layout area and from the Logging Display drop-down list box, choose the applicable profile.

The Disabled option is selected by default.

Parameters that you set in the Product Specific Configuration area may also appear in the Device Configuration window for various devices and in the Enterprise Phone Configuration window.

If you set these same parameters in these other windows as well, the setting that takes precedence is determined in the following order:

a. Device Configuration window settings

b. Common Phone Profile window settings

c. Enterprise Phone Configuration window settings

Services Setup

You can give users access to Cisco Unified IP Phone Services on the Cisco Unified IP Conference Phone. These services comprise XML applications that enable the display of interactive content with text and graphics on the conference phone. Examples of services include local movie times, stock quotes, and weather reports. Users access any configured services from the **Apps** softkey on the conference phone.

Before a user can access any service:

- You must use Cisco Unified Communications Manager Administration to configure available services.
- The user must subscribe to services by using the Cisco Unified Communications Self Care Portal. This web-based application provides a graphical user interface (GUI) for limited, end-user configuration of IP phone applications. However, a user cannot subscribe to any service that you configure as an enterprise subscription.

Before you set up services, gather the URLs for the sites you want to set up, and verify that users can access those sites from your corporate IP telephony network.

To set up these services choose **Device** > **Device Settings** > **Phone Services** from Cisco Unified Communications Manager. For more information, see the *Cisco Unified Communications Manager Administration Guide* and *Cisco Unified Communications Manager System Guide*.

After you configure these services, verify that your users have access to Cisco Unified Communications Manager Self Care Portal, from which they can select and subscribe to configured services. See IP Phone Features User Subscription and Setup for a summary of the information that you must provide to end users.

Note To configure Cisco Extension Mobility services for users, see "Cisco Unified Mobility", *Cisco Unified Communications Manager Features and Services Guide*

Cisco Unified Communications Manager User Addition

Adding users to Cisco Unified Communications Manager allows you to display and maintain information about users and allows each user to perform these tasks:

- Access the corporate directory and other customized directories from a Cisco Unified IP Phone.
- Create a personal directory.
- Set up Call Forwarding numbers.
- Subscribe to services that are accessible from a Cisco Unified IP Phone.

You can add users to Cisco Unified Communications Manager using either of these methods:

- To add users individually, choose **User Management** > **End User** from Cisco Unified Communications Manager Administration.

  For more information on adding users, see the *Cisco Unified Communications Manager Administration Guide*. For details on user information, see the *Cisco Unified Communications Manager System Guide*.

- To add users in batches, use the Bulk Administration Tool. This method also enables you to set an identical default password for all users.

  For more information, see the *Cisco Unified Communications Manager Bulk Administration Guide*.

- To add users from your corporate LDAP directory, choose **System** > **LDAP** > **LDAP System** from Cisco Unified Communications Manager Administration.

  Note After the Enable Synchronization from LDAP Server is enabled, you will not be able to add additional users from Cisco Unified Communications Manager Administration.

  For more information on LDAP, see the *Cisco Unified Communications Manager System Guide*, “Understanding the Directory”.

- To add a user and a phone at the same time, choose **User Management** > **User/Phone Add** from Cisco Unified Communications Manager.

Call Waiting Setup

The Cisco Unified IP Conference Phone supports six calls on a single line. With multiple calls per line, setting up call waiting is simplified on the Cisco Unified Communications Manager. For more information, see “Understanding Directory Numbers”, *Cisco Unified Communications Manager System Guide*.

Call Forward Notification Setup

You set up the information that is displayed to the user from within Cisco Unified Communications Manager Administration in the Device Configuration window (**Device** > **Phone** > **Line** > **Forwarded Call Information Display on Device**).

The following table describes the Call Forward Notification fields.

Table 4: Call Forward Notification Fields

Field	Description	Default Checkbox State
Caller Name	When this check box is checked, the caller name displays in the notification window.	Checked
Caller Number	When this check box is checked, the caller number displays in the notification window.	Not checked

Field	Description	Default Checkbox State
Redirected Number	When this check box is checked, the information about the caller who last forwarded the call displays in the notification window. Example: If Caller A calls B, but B has forwarded all calls to C and C has forwarded all calls to D, the notification box that D sees contains the phone information for caller C.	Not checked
Dialed Number	When this check box is checked, the information about the original recipient of the call displays in the notification window. Example: If Caller A calls B, but B has forwarded all calls to C and C has forwarded all calls to D, then the notification box that D sees contains the phone information for caller B.	Checked

Calling Party Normalization

In line with E.164 standards, 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 having to modify the directory number in the call log directories on the phone. Additionally, calling party normalization allows the user to globalize and localize phone numbers, so the appropriate calling number presentation displays on the phone.

The conference phone supports the following functions:

- For the final presentation of the calling number to the user, the conference phone screen displays the calling number based on the international, national, or local subscriber numbers.
    - If the call is an intra-city call, the calling number presented on the conference phone is displayed in the subscriber number format (without the area or city code).
    - For intercity calls, the calling number is presented in a national number format.
    - If the call is an international call, the calling number is presented with the E.164 format, with the plus (+) prefix digit.
- The call logs directories record the calling number in the received and missed call logs with the appropriate escape codes (9/0, 0/1, +). The user can go into directories, and select and dial one of these entries with the escape code without having to edit the number.

Configuring calling party normalization alleviates issues with toll bypass where the call is routed to multiple locations over the IP WAN. In addition, it allows Cisco Unified Communications Manager to distinguish the origin of the call to globalize or localize the calling party number for the conference phone user.

The conference phone itself can localize the calling party number. For the phone to localize the calling party number, you must configure the Calling Party Transformation CSS or the Use Device Pool Device Calling Party Transformation CSS setting in the Phone Configuration window.

Depending on your configuration for globalizing and localizing the calling party number, the phone user may see a localized number, a globalized number with access codes and prefixes, or the international escape character, +, in the calling party number. If a phone supports calling party normalization, the phone can show the localized calling party number on the conference phone screen and the globalized number in the call log directories on the conference phone.

In addition, these devices show both the globalized and localized calling party number in the Call Details. If a conference phone does not support calling party normalization, the device shows the localized calling party on the conference phone screen and in the call log directories on the phone.

For information on how to configure this feature for your conference phone, see "Calling Party Normalization", *Cisco Unified Communications Manager Features and Services Guide*.

Incoming Call Notification Window Timer Setup

You can set the time that the Incoming Call Notification Window, sometimes called a toast, displays on the conference phone. You set up the feature from one of the following Cisco Unified Communications Manager windows:

- Enterprise Phone Configuration (**System** > **Enterprise Phone**)
- Common Phone Profile Configuration (**Device** > **Device Settings** > **Common Phone Profile**)
- Phone Configuration (**Device** > **Phone**)

The following table describes the Incoming Call Toast Timer.

Table 5: Incoming Call Notification Window Timer Field

Field	Description	Default Value
Incoming Call Toast Timer	Gives the time, in seconds, that the toast displays. The time includes the fade-in and fade-out times for the window. The possible values are 3, 4, 5, 6, 7, 8, 9, 10, 15, 30, and 60.	5 seconds.

Linked Mode

Two conference phone Sound Base units can be linked together to expand the audio coverage area. One Sound Base acts as the primary device, and the other unit acts as the dependant or secondary device.

In Linked Mode, the primary base station supports either one wireless or one wired microphone. The secondary unit supports only one wired microphone. You cannot mix microphone kits: if you plan to connect a microphone to both sound bases, they must both be wired microphones.

The Cisco Unified IP Conference Phone 8831 supports wired and wireless microphones. The Cisco Unified IP Conference Phone 8831NR supports only wired microphones.

The voice, dial tone, ringer, and base LED features synchronize between the two devices when linked together. You can link two sound bases while a call is active.

When Linked Mode is active, the linked mode icon displays in the idle and the call screens.

The following table summarizes the best practice to follow when deploying your conference phones in Linked Mode. If the devices are linked in this manner, the system software automatically detects which device is to be used as the primary and which is the secondary one.

Table 6: Linked Mode Setup Best Practice

Component	Connect to Primary	Connect to Secondary
Display Control Unit (DCU)	Yes	No
Network cable	Yes	No
Wall power	Yes	No
Optional Wired Microphone	Yes	Yes
Optional Wireless Microphone	Yes	No

Note If a DCU is connected to the secondary device, it will display a prompt indicating that it is a dummy DCU, but will otherwise not function.

Caution When using a Sound Base in Linked Mode, the primary base unit must be connected using the CP-PWR-CUBE-3 external power supply.

If two devices are linked after both are registered, the user can select which is the primary device.

A secondary device receives upgrades to firmware seamlessly from the primary device.

Link Conference Phones

Use a daisy cable to connect two sound base units in Linked mode. This procedure describes the best practice for connecting the two units.

Procedure

Step 1 Connect the DCU to the conference phone to be used as the primary unit.

Step 2 Connect the network cable to the conference phone to be used as the primary unit.

Step 3 Connect the power cable to the primary device and plug into a wall plug.

The secondary Sound Base does not need to be plugged into external power, but in Linked Mode the primary unit must be connected to external power.

Step 4 Use the provided daisy cable to connect the primary unit to the secondary sound base.

Voice, dial tone, ringer and base LEDs synchronize between the two units.

Related Topics

Linked Mode, on page 26