Cisco Unified IP Conference Phone 8831 Release Notes for Firmware Release 9.3(3)

First Published: 2013-06-21

Last Modified: 2020-07-02

CONTENTS

Cisco Unified IP Conference Phone 8831 Release Notes for Firmware Release 9.3(3)

Introduction

These release notes support Cisco Unified IP Conference Phone 8831 running SIP firmware Release 9.3(3).

The following table lists the Cisco Unified Communications Manager release and protocol compatibility for the Cisco Unified IP Phones.

Table 1: Cisco Unified IP Phones, Cisco Unified Communications Manager, and Firmware Release Compatibility

Cisco Unified IP Phone	Protocol	Cisco Unified Communications Manager
Cisco Unified IP Conference Phone 8831	SIP	Cisco Unified Communications Manager Release 7.1.5 and later

Note SIP Firmware Release 9.3(3) is designed and tested to interoperate with Cisco call control, most notably Cisco Unified Communications Manager Release 9.0(1). Although SIP firmware is IETF RFC 3261 compliant, it is not supported by Cisco TAC or Engineering for use with non-Cisco call control systems.

Cisco Unified IP Conference Phone 8831

The Cisco Unified IP Conference Phone 8831 is a full-featured single line conference station that provides voice communication over an IP network. It functions much like a digital business phone, allowing you to place and receive calls and to access features such as mute, hold, transfer, speed dial, call forward, and more. In addition, because conference stations connect to your data network, they offer enhanced IP telephony features, including access to network information and services, and customizable features and services.

The conference phone provides a backlit LCD screen, support for up to ten speed-dial numbers, and a variety of other sophisticated functions. Optional microphone extension kits provide enhanced room coverage that can be further expanded by linking two units together.

In addition to basic call handling features, your conference phone can provide enhanced productivity features that extend your call handling capabilities. Depending on configuration, the phone supports:

- Access to network data, XML applications, and web-based services.

- Online customizing of conference station features and services from the User Options web pages.

The phone supports seamless firmware upgrade. If two devices are connected in Linked Mode, firmware upgrades are pushed automatically from the primary unit to the secondary unit.

For information on deploying the conference phone, see *Cisco Unified IP Conference Phone 8831 Administration Guide for Cisco Unified Communications Manager*, "Cisco Unified IP Conference Phone 8831 setup".

Language Support

The default language for Cisco Unified IP Conference Phone 8831 with Firmware Release 9.3(3) is American English. For other locales, if the language does not support certain words and phrases, the phone displays the English phrase.

Cisco provides locale-specific versions of the Cisco Unified Communications Manager Locale Installer on www.cisco.com. Installed by the system administrator, the locale installer allows the user to view and receive the chosen translated text or tones, if applicable, when a user works with supported interfaces.

User Locales
>   User locale files provide translated text and voice prompts, if available, for phone displays, user applications, and user web pages in the locale that the user chooses. User-only locale installers exist on the web.

Network Locales
>   Network locale files provide country-specific phone tones and gateway tones, if available. Network-only locale installers exist on the web.

Cisco may combine multiple network locales in a single locale installer.

For more information on international user support, see *Cisco Unified IP Conference Phone 8831 Administration Guide*, "Appendix B", International user support.

For more information on installing locales, see *Cisco Unified Communications Operating System Administration Guide, Release 9.1(1)*, "Software upgrades", Locale installation.

Related Documentation

Use the following sections to obtain related information.

Cisco IP Phone 8800 Series Documentation

Refer to publications that are specific to your language, phone model, and call control system. Navigate from the following documentation URL:

https://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-8800-series/index.html

The Deployment Guide is located at the following URL:

https://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8800-series/products-implementation-design-guides-list.html

Cisco Unified Communications Manager Documentation

See the *Cisco Unified Communications Manager Documentation Guide* and other publications that are specific to your Cisco Unified Communications Manager release. Navigate from the following documentation URL:

https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/tsd-products-support-series-home.html

Cisco Business Edition 6000 Documentation

Refer to the *Cisco Business Edition 6000 Documentation Guide* and other publications that are specific to your Cisco Business Edition 6000 release. Navigate from the following URL:

https://www.cisco.com/c/en/us/support/unified-communications/business-edition-6000/tsd-products-support-series-home.html

Features Available with Firmware Release 9.3(3)

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 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 2: 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.
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.

Feature	Description	Configuration Reference
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"
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.

Feature	Description	Configuration Reference
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.
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.

Feature	Description	Configuration Reference
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
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"

Feature	Description	Configuration Reference
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"
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 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.

Feature	Description	Configuration Reference
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.
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.

Feature	Description	Configuration Reference
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.
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"
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.
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.)

Feature	Description	Configuration Reference
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 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"
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.
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.

Feature	Description	Configuration Reference
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>.
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"

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 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 3: 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.

Installation

Installation Requirements

Before you install the firmware release, you must ensure that your Cisco Unified Communications Manager is running the latest device pack. The applicable device packs are released after the firmware release. After you install a device pack on the Cisco Unified Communications Manager servers in the cluster, you need to reboot all the servers.

Note If your Cisco Unified Communications Manager does not have the required device pack to support this firmware release, the firmware may not work correctly.

For information on the Cisco Unified Communications Manager Device Packs, see http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/compat/devpack_comp_mtx.html.

Install Latest Cisco Unified Communications Manager Release

Before using the Cisco Unified IP Phone with Cisco Unified Communications Manager, your Cisco Unified Communications Manager servers must be running a version of the server software that supports the phones. All Cisco Unified Communications Manager servers in the cluster must support the phones. For information about the minimum Cisco Unified Communications Manager software version that the phone requires, see the introductory sections of these release notes.

For more information on Cisco Unified Communications Manager installations and upgrades, see the documents for your Cisco Unified Communications Manager version at the following location: http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_installation_guides_list.html

To download and install the Cisco Unified Communications Manager version, perform these steps.

Procedure

Step 1 Go to the following URL:

http://www.cisco.com/cisco/software/navigator.html?mdfid=268439621&catid=278875240

Step 2 Choose your Cisco Unified Communications Manager version.

Step 3 Choose the appropriate software type.

Step 4 Hover over the desired file. When the popup window displays, click the **Readme** link to open the readme file.

Step 5 Choose **Download** or **Add to cart** for the desired file.

Step 6 Use the instructions in the readme file to install the updated file on the Cisco Unified Communications Manager.

Install Cisco Unified Communications Manager Device Packs

Device packs are required to enable IP phones in the Cisco Unified Communications Manager database. For information about compatible device packs, see http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/compat/devpack_comp_mtx.html.

For Cisco Unified Communications Manager 6.0 and later, you must run the device pack and reboot the Cisco Unified Communications Manager server.

Procedure

Step 1 Go to the following URL: .

http://www.cisco.com/cisco/software/navigator.html?mdfid=268439621&flowid=21301

Step 2   Choose your Cisco Unified Communications Manager version.

Step 3   Hover over the desired device pack. When the popup window displays, click the **Readme** link to open the readme file.

Step 4   Choose **Download** or **Add to cart** for the desired device pack.

Step 5   Use the instructions in the readme file to install the updated device pack on the Cisco Unified Communications Manager.

Install Firmware Release 9.3(3) on Cisco Unified Communications Manager

Before you begin

Before using Cisco Unified IP Phone Firmware Release 9.3(3) with Cisco Unified Communications Manager, you must install the latest firmware on all Cisco Unified Communications Manager servers in the cluster.

Procedure

Step 1   Go to the following URL:

http://www.cisco.com/cisco/software/navigator.html?mdfid=268437892&flowid=5293

Step 2   Choose **Cisco Unified IP Phones 8800 Series**.

Step 3   Choose your phone.

Step 4   Choose **Session Initiation Protocol (SIP) Software**.

Step 5   In the Latest Releases folder, choose **9.3(3)**.

Step 6   Select one of the following firmware files, click the **Download Now** or **Add to cart** button, and follow the prompts:

```
cmterm-8831-sip.9-3-3-5.cop.sgn
```

Note   If you added the firmware file to the cart, click the **Download Cart** link when you are ready to download the file.

Step 7   Click the + next to the firmware file name in the Download Cart section to access additional information about this file. The hyperlink for the readme file is in the Additional Information section, which contains installation instructions for the corresponding firmware:

```
cmterm-8831-sip.9-3-3-5-readme.html
```

Step 8   Follow the instructions in the readme file to install the firmware.

Install Firmware Zip Files

If a Cisco Unified Communications Manager is not available to load the installer, the following .zip file is available to load the firmware:

- `cmterm-8831-sip.9-3-3-5.zip`

Procedure

Step 1   Go to the following URL:

http://www.cisco.com/cisco/software/navigator.html?mdfid=268437892&flowid=5293

Step 2   Choose **Cisco Unified IP Phones 8800 Series**.

Step 3   Choose your phone type.

Step 4   Choose **Session Initiation Protocol (SIP) Software**.

Step 5   In the Latest Releases folder, choose **9.3(3)**.

Step 6   Download the relevant zip files.

Step 7   Unzip the files.

Step 8   Manually copy the unzipped files to the directory on the TFTP server. See *Cisco Unified Communications Operating System Administration Guide* for information about how to manually copy the firmware files to the server.

Install Firmware on Phone

Procedure

Step 1   Download the latest firmware load.

`cmterm-8831-sip.9-3-3-5.cop.sgn`

Step 2   Login to the Cisco Unified Operating System Administration web page from your web browser.

Step 3   Select Install/Upgrade from the Software Upgrades menu.

Step 4   Enter the location values in the Software Location fields for the file.

Step 5   Select the file from the Options/Upgrades drop-down list.

Step 6   Click Next and follow the prompts.

Step 7   Restart the TFTP service on Cisco Unified Communications Manager.

    a) Login to the Cisco Unified Serviceability web page from your web browser.

    b) Select Control Center - Feature Services from the Tools menu.

    c) Click the Cisco TFTP radio button in the CM Services section.

    d) Click the restart icon at the top of the page.

    The TFTP server restarts.

Seamless Firmware Upgrade

The conference phone supports seamless firmware upgrade. Once the firmware is installed on the base unit and it completes its reboot sequence, the firmware on the display control unit (DCU) upgrades automatically via USB connection. No reboot of the base unit is required after upgrading the firmware on the DCU.

If an optional wireless microphone extension kit is linked to the sound base, its firmware is also automatically upgraded. The wireless microphone performs a reboot after a firmware upgrade, but no reboot of the base unit is required.

If two devices are connected in Linked Mode, firmware upgrades are pushed automatically from the primary unit to the secondary unit.

Note   The device pack for Firmware Release 9.3(3) must be installed on the Cisco Unified Communications Manager prior to firmware upgrade.

Upgrade Sound Base Firmware

The following provides an overview of the steps in the upgrade process.

Procedure

Step 1   Install the latest version of the firmware load on Cisco Unified Communications Manager.

```
cmterm-8831-sip.9-3-3-5.cop.sgn
```

Step 2   Restart the TFTP service on Cisco Unified Communications Manager.

Step 3   Wait for the phone to reboot.

The firmware upgrade is applied and the phone reboots. Your conference phone is ready for use.

Display Control Unit Firmware Upgrade

The conference phone supports seamless firmware upgrade. Once the firmware is installed on the Sound Base and it completes its reboot sequence, the firmware on the display DCU upgrades automatically over the USB connection to the Sound Base. No reboot of the base unit is required after upgrading the firmware on the DCU.

Wireless Microphone Firmware Upgrade

If an optional wireless microphone extension kit is linked to the Sound Base, its firmware is automatically upgraded after the base unit upgrade is completed. The wireless microphone performs a reboot after a firmware upgrade, but no reboot of the base unit is required.

Limitations and Restrictions

Wireless Microphone Region Setting

The default value for the Wireless Microphone Region setting is United States. If you use the device in a locale outside the United States and you configure a wireless microphone to pair with the device, you must change the Wireless Microphone Region setting to your region. This requirement ensures that the wireless microphone does not interfere with local bandwidth usage.

Use Cisco Unified Communications Manager Administration to change the Wireless Microphone Region setting.

Set Wireless Microphone Region

Procedure

Step 1   In Cisco Unified Communications Manager Administration, go to **Device** > **Phone**.

Step 2   Use the Find capability to find the Cisco Unified IP Conference Phone 8831.

Step 3   In the Product Specific Configuration Layout area of the Phone Configuration window, change the Wireless Microphone Region setting to the region where the device is located. The drop-down list box lists the available regions.

Phone Behavior During Times of Network Congestion

Anything that degrades network performance can affect phone audio and, in some cases, can cause a call to drop. Sources of network degradation can include, but are not limited to, the following activities:

- Administrative tasks, such as an internal port scan or security scan

- Attacks that occur on your network, such as a Denial of Service attack

Unified Communications Manager Endpoints Locale Installer

By default, Cisco IP Phones are set up for the English (United States) locale. To use the Cisco IP Phones in other locales, you must install the locale-specific version of the Unified Communications Manager Endpoints Locale Installer on every Cisco Unified Communications Manager server in the cluster. The Locale Installer installs the latest translated text for the phone user interface and country-specific phone tones on your system so that they are available for the Cisco IP Phones.

To access the Locale Installer required for a release, access https://software.cisco.com/download/navigator.html?mdfid=286037605&flowid=46245, navigate to your phone model, and select the Unified Communications Manager Endpoints Locale Installer link.

For more information, see the documentation for your particular Cisco Unified Communications Manager release.

Note   The latest Locale Installer may not be immediately available; continue to check the website for updates.

Caveats

This section describes the resolved and open caveats, and provides information on accessing the Cisco Software Bug Toolkit.

Access Cisco Bug Search

Known problems (bugs) are graded according to severity level. These release notes contain descriptions of the following:

- All severity level 1 or 2 bugs

- Significant severity level 3 bugs

You can search for problems by using Cisco Bug Search.

Before you begin

To access Cisco Bug Search, you need the following items:

- Internet connection

- Web browser

- Cisco.com user ID and password

Procedure

Step 1   To access Cisco Bug Search, go to:

https://tools.cisco.com/bugsearch

Step 2   Log in with your Cisco.com user ID and password.

Step 3   To look for information about a specific problem, enter the bug ID number in the Search for field, then press **Enter**.

Open Caveats

The following table lists severity 1, 2, and 3 defects that are open for Cisco Unified IP Conference Phone 8831 with Firmware Release 9.3(3).

For more information about an individual defect, you can access the online record for the defect by clicking the Identifier or going to the URL that is shown. You must be a registered Cisco.com user to access this online information.

Because defect status continually changes, the table reflects a snapshot of the defects that were open at the time this report was compiled. For an updated view of open defects, access Bug Toolkit as described in .

Table 4: Open Caveats for Firmware Release 9.3(3)

Identifier	Headline
CSCtz98863	"Redirected" and "Dialed" Number not displayed when receiving CFwd call
CSCuc36050	Phone resets twice on performing "reset all"
CSCuc72874	Clicking noise is heard when pressing mute button
CSCud18915	Issue with start record when it is a supervisor monitoring call
CSCud47672	The LEDs on Base and DCU do not turn on when RTPTx and RTPMTx
CSCud49802	8831 does not play the sound of RTPRx when ring in
CSCud99174	"Not Registered" message always shows while setting up phone through DCU
CSCue01254	Locale: local issues for Arabic
CSCue03730	Misc. locale formatting and translation issues on 8831
CSCue09847	Locale: Network locale summary issues
CSCue39992	Locale: Chosen Ringtone name not translated when changing locale
CSCue42487	Feature "Calling Party Normalization" is broken
CSCue78002	Locale: In HK, Call 2 of 2 is not translated correctly during call
CSCuf52937	LM upgrade failed due to no reboot command from primary
CSCug20247	Can't conf another conference via Calls softkey

Cisco IP Phone Firmware Support Policy

For information on the support policy for phones, see https://cisco.com/go/phonefirmwaresupport.

Documentation, Service Requests, and Additional Information

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/c/en/us/td/docs/general/whatsnew/whatsnew.html

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS Version 2.0.

INDEX

B

bug **23**

D

DECT **22**

W

Wireless Microphone **22**
   DECT **22**
   Region Setting **22**
   Set Region **22**