Cisco Unified IP Conference Phone 8831 User Guide for Cisco Unified Communications Manager 9.0

First Published: June 21, 2013

Last Modified: March 26, 2015

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CHAPTER 1

Preface

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- Additional Information, page 2
- Safety and Performance Information, page 2
- Accessibility Features, page 3

Introduction

This guide provides you with an overview of the features available on your Cisco Unified IP Conference Phone 8831. You can read it completely for a solid understanding of the capabilities of your phone or refer to the table below for pointers to commonly used sections.

If You Want to...	Then...
Review safety information	See Safety and Performance Information, on page 2.
Connect the phone	See Phone Connections, on page 15.
Use the phone after it is installed	Start with Features of Your Cisco Unified IP Conference Phone, on page 5.
Learn what the phone buttons mean	See Buttons and Hardware, on page 5.
Learn about the display screen	See Phone Screen, on page 13.
Make calls	See Calling Features, on page 37.
Put calls on hold	See Hold, on page 51.
Mute calls	See Mute, on page 57.
Transfer calls	See Transfer, on page 61.
Make conference calls	See Conference, on page 45.

If You Want to...	Then...
Set up speed dialing	See Speed Dial, on page 61.
Share a phone number	See Shared Lines, on page 60.
Change the ring volume or type	See Ringtones, on page 22.
View your missed calls	See Call History, on page 19.
Listen to your voice messages	See Messages, on page 35.

Additional Information

You can access the most current Cisco documentation at this URL:

http://www.cisco.com/en/US/products/ps12965/tsd_products_support_series_home.html

You can access the Cisco website at this URL:

http://www.cisco.com/

You can access the most current licensing information at this URL:

http://www.cisco.com/en/US/products/ps12965/products_licensing_information_listing.html

Safety and Performance Information

The following sections provide information about the impact of power outages and external devices on your Cisco Unified IP Phone.

Power Outage

Your access to emergency service through the phone depends on the phone being powered. If there is a power interruption, Service and Emergency Calling Service dialing will not function until power is restored. In case of a power failure or disruption, you may need to reset or reconfigure the equipment before using the Service or Emergency Calling Service dialing.

External Devices

Cisco recommends using good quality external devices such as headsets, cables, and connectors that are shielded against unwanted radio frequency (RF) and audio frequency (AF) signals.

Note Not all Cisco IP Telephony products support external devices, cords or cables. For more information, consult the documentation for your phone.

Depending on the quality of these devices and their proximity to other devices such as mobile phones or two-way radios, some audio noise may still occur. In these cases, Cisco recommends that you take one or more of these actions:

- Move the external device away from the source of the RF or AF signals.
- Route the external device cables away from the source of the RF or AF signals.
- Use shielded cables for the external device, or use cables with a better shield and connector.
- Shorten the length of the external device cable.
- Apply ferrites or other such devices on the cables for the external device.

Cisco cannot guarantee the performance of external devices, cables, and connectors.

Caution In European Union countries, use only external speakers, microphones, and headsets that are fully compliant with the EMC Directive [89/336/EC].

Cisco Product Security Overview

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer, and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute, or use encryption. Importers, exporters, distributors, and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

Further information regarding U.S. export regulations may be found at http://www.bis.doc.gov/index.php/regulations/export-administration-regulations-ear.

Phone Behavior During Times of Network Congestion

Anything that degrades network performance can affect Cisco IP Phone voice and video quality, 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

To reduce or eliminate any adverse effects to the phones, schedule administrative network tasks during a time when the phones are not being used or exclude the phones from testing.

Accessibility Features

The Cisco Unified IP Conference Phone 8831 provides accessibility features for the vision impaired, the blind, and the hearing and mobility impaired.

For detailed information about the features on these phones, see http://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8800-series/products-technical-reference-list.html.

You can also find more information about accessibility at this Cisco website:

http://www.cisco.com/web/about/responsibility/accessibility/index.html

CHAPTER 2

Features of Your Cisco Unified IP Conference Phone

- Features Overview, page 5
- Buttons and Hardware, page 5
- Phone Screen, page 13
- Phone Connections, page 15
- General IP Phone Information, page 18
- Feature Availability, page 18

Features Overview

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, and call forward. In addition, because conference stations connect to your data network, they offer enhanced IP telephony features, including access to network information, network services, and customizable features and services.

The conference station 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 station can provide enhanced productivity features that extend your call-handling capabilities. Depending on configuration, the conference station supports:

- Access to network data, XML applications, and web-based services.
- Online customizing of conference station features and services from your User Options web pages.

Buttons and Hardware

The Conference Phone has two primary components:

- Display Control Unit (DCU)
- Sound Base

In addition, the following optional extension kits can be added to or used with the conference phone:

- Wired Microphone Extension Kit
- Wireless Microphone Extension Kit and Charger

For your conference phone to work, it must be connected to the corporate IP telephony network.

LED State Definitions

LEDs on the Sound Base and DCU provide information about the state of the conference phone.

For example, green flashing lights on the Sound Base and on the DCU Call button indicate that there is an incoming call. If the conference phone is on mute, then an incoming call will still flash green on the Call button, but the LED for the DCU mute button is solid red, the sound base LEDs are solid red, and the mute button on the Sound Base is also solid red.

The following table is a guide to the behaviour of the LEDs on the sound base and the DCU.

Table 1: Conference Phone LED State Table

Media Path Status	Call on Focus	Sound Base				Display Control Unit (DCU)			
		Base LEDs (3)		Mute Button		DCU Call Button		DCU Mute Button	
Off	No call								
Off	No call, with VM					red	solid		
Off	DND flash	green	flash			green	flash		
Off	Incoming call	green	flash			green	flash		
Off	Hold Revert Call	green	flash			green	flash		
Off	Hold Call	green	pulse			green	pulse		
Off	Hold Remote Call					red	pulse		
Off	Remote in use Call					red	solid		
Unmuted	Ringout/Connected Call	green	solid			green	solid		
Unmuted	DNDFlash	green	solid			green	flash		
Unmuted	Incoming Call	green	solid			green	flash		
Unmuted	Hold Revert Call	green	solid			green	flash		

Media Path Status	Call on Focus	Sound Base				Display Control Unit (DCU)			
		Base LEDs (3)		Mute Button		DCU Call Button		DCU Mute Button	
Muted	Ringout/Connected	red	solid	red	solid	green	solid	red	solid
Muted	DND Flash	red	solid	red	solid	green	flash	red	solid
Muted	Incoming Call	red	solid	red	solid	green	flash	red	solid
Muted	Hold Revert Call	red	solid	red	solid	green	flash	red	solid
Deep Sleep Mode	Deep Sleep Mode			gray	solid				

Display Control Unit

The Display Control Unit (DCU) is tethered to the Sound Base via a micro USB connector.

You can use the graphic and table below to identify buttons and hardware on the DCU.

Table 2: Display Control Unit Buttons and Softkeys

	Item	Description
1	Phone screen	LCD screen that displays conference phone menus and features.
2	Softkeys	Four programmable keys.
3	Navigation bar with Select key	2-way Navigation bar and Select key that allows you to scroll menus and select items on the display.
4	Call button	LED backlit call button. Press this key to: • Go Off Hook • Answer an incoming call • Obtain a dial tone to initiate a call • Resume a call • Release a call
5	Keypad	Allows you to dial phone numbers and enter letters.
6	Mute button	Toggles the Mute feature. A red backlight indicates a call is on mute.
7	Volume rocker	2-way rocker switch that raises the volume of the speaker.

Note For details on DCU LED behavior, see LED State Definitions, on page 6.

Sound Base

The Sound Base provides 360 degree audio coverage via four built-in microphones and supports a full duplex speaker phone.

To provide enhanced room coverage, two sound base units can be linked together.

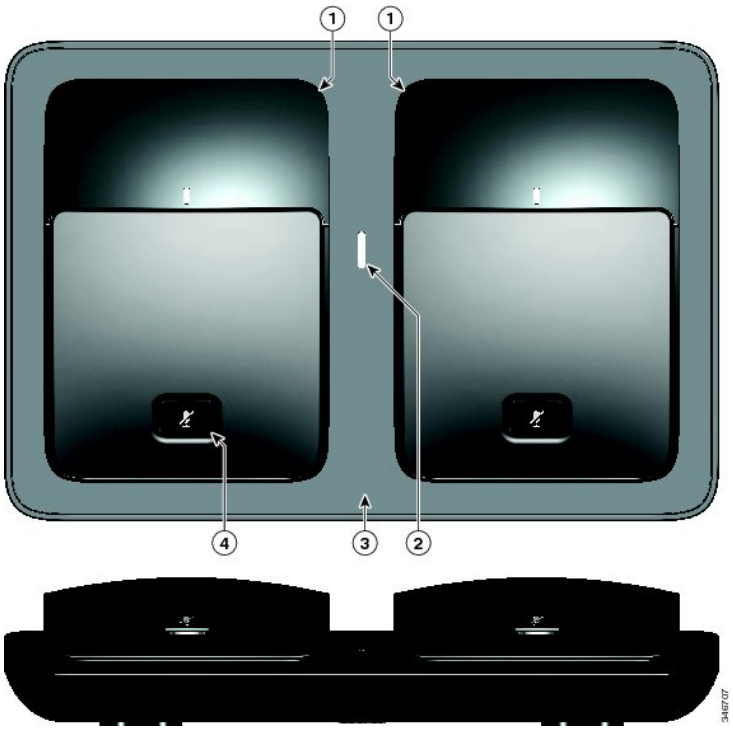
You can use the graphic and table below to identify buttons and connections on the Sound Base.

Table 3: Sound Base Buttons

	Item	Description
1, 2, 3	LED indicators	Three LED indicators provide call status information. For details on LED behaviour, see LED State Definitions, on page 6.
4	Mute button	Backlit mute button.

Wireless Extension Microphone and Charger Kit

The optional wireless extension microphone and charger kit unit contains two omni-directional wireless microphones, a charger unit, and two country-specific AC power clip adaptors for powering the charger unit. Connecting a microphone kit enhances the room coverage of the conference station.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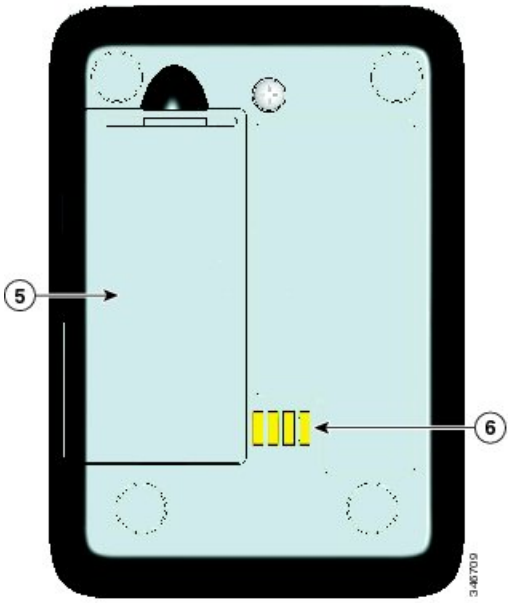


Table 4: Wireless Microphone and Charger Buttons and Features

Item	Description
1	Omni-directional wireless microphones
2	LED charge status indicator
3	Charger tray
4	Mute button
5	Battery compartment
6	Charging contacts

Note See the FCC ID approval number under the Battery Cover for CP-MIC-WRLS-S-US= Wireless Microphone. FCC ID: LDKMICWR0776

Note See the Canada IC approval number under the Battery Cover for the CP-MIC-WRLS-S-US= Wireless Microphone. IC: 2461B-MICWR0776

Wired Extension Microphone Kit

The optional wired extension microphone kit includes two wired omni-directional microphones. Connecting a microphone kit enhances the room coverage of the conference phone. The sound base has two wired microphone ports and you can connect one or both wired microphones.

If the conference phone is connected to another sound base in Linked Mode, the primary base station supports one or two wireless microphones, or it supports one wired microphone. The secondary unit supports only one wired microphone; a wireless microphone cannot be connected to a secondary Sound Base. You cannot mix microphone kits: if you plan to connect a microphone to both sound bases, they must both be wired microphones.

Note Wired and wireless microphones cannot be used at the same time, and the wireless microphones have a higher priority. Attempting to connect a wired microphone to a conference phone that has paired or connected channels results in a warning to the user that the wired microphone is disabled. To solve this problem, unpair any paired or connected wireless microphones before connecting a wired microphone.

Table 5: Wired Microphone Buttons

Item	Description
1	Mute button.

Phone Screen

The DCU contains the LCD phone screen. The idle or home screen displays information about the status of calls and features.

If the conference phone is in an offline state, the idle screen displays the message `Phone is not registered` and the **Apps** softkey remains available.

You can use the graphic and table below to identify the features and functions available on the screen.

Table 6: Phone Screen Layout.

	Item	Description
1	Header	Displays date, time, and current directory number. Displays menu name when applicable.
2	Line details and other phone information	Displays line label, call details, and status messages such as missed calls, message waiting, and line forwarding information.
3	Call State icon	Indicates the status of a call, such as ringing, hold, encrypted or connected call.
4	Softkey labels	Displays softkeys for currently available features or actions.
5, 6	Feature icons	These icons are displayed when an associated feature, such as extension microphones (5) or Link mode (6) is connected.

Phone Screen Icons

Table 7: Phone Screen Icons

Icon	Description
	On hook
	Off hook
	Ringing in
	Connected
	Hold
	Shared line
	Microphone connected
	Linked mode

Icon	Description
🔒	Encrypted

Phone Screen Cleaning

Caution Use only a soft, dry cloth to wipe your conference station display screen. Do not use any liquids or powders on the phone, as they can contaminate phone components and cause failures.

Phone Connections

The Sound Base contains the network and power connection for the phone. The base also contains the mini-USB connection for the DCS, the wired microphone ports, and the daisy chain port for the Linked Mode feature.

You can use the graphic and table below to identify connections and ports on the Sound Base.

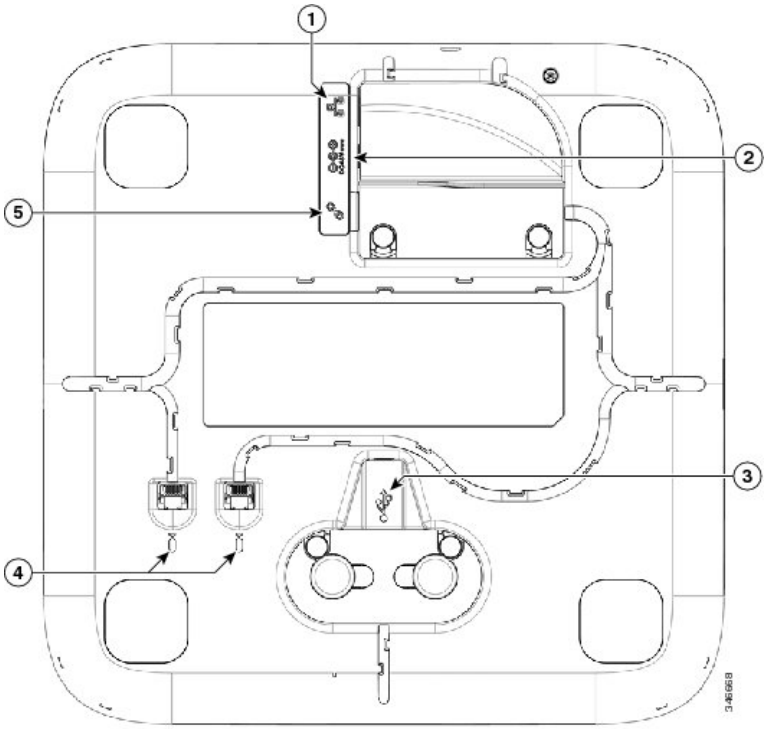
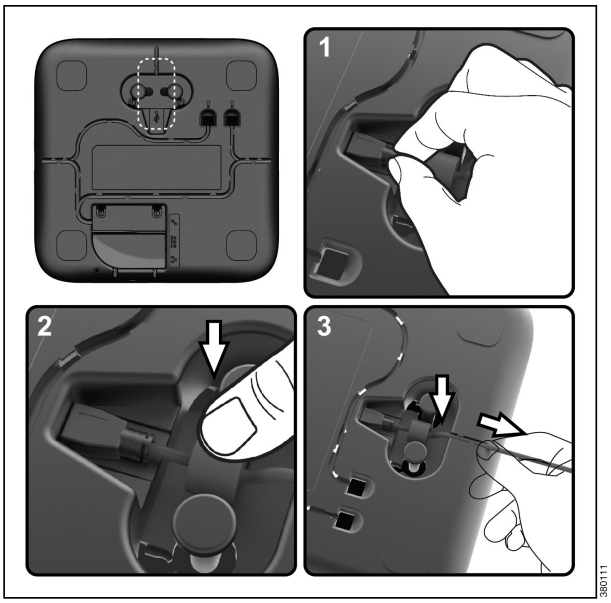


Table 8: Sound Base Connections and Ports

	Item	Description
1	Network port	Network port (10/100 SW) connection. IEEE 802.3af power enabled.
2	Wall power	Local power connection.
3	Mini USB port	Connects the base station to the DCU. **Attention** When connecting the USB cable, firmly press down on the ferrite bead to ensure it seats correctly between the posts.
4	Wired microphone ports	Two RJ11 microphone ports. An optional wired microphone can be connected to each port.
5	Linked Mode daisy chain port	Supports the connection of two base stations in Linked Mode.

Connect Mini-USB Cable to DCU

The following images show how to install the USB cable.



Procedure

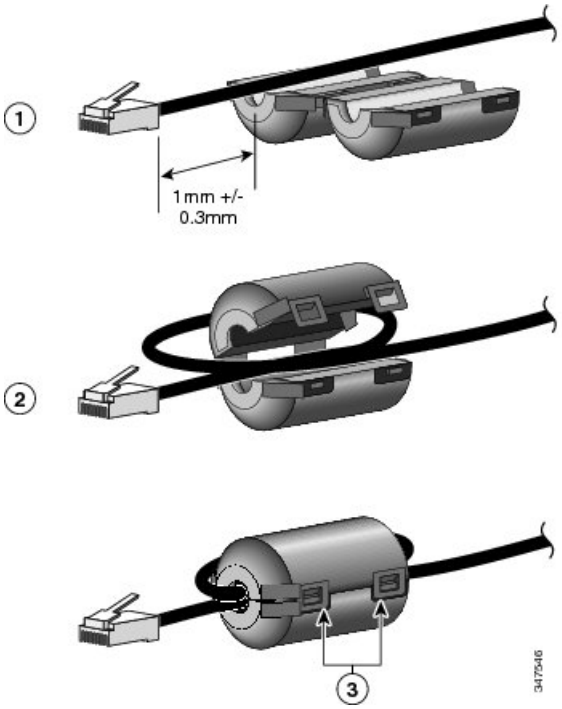
Step 1 Insert the mini-USB connector into the port on the base of the phone.

Step 2 Seat the ferrite bead between the posts and press the bead down firmly.

Step 3 Thread the cable in the cable channel. Make sure that you leave some slack in the cable.

Install Ferrite Bead on Network Cable

The following images show how to install the ferrite bead on the network cable.



Procedure

Step 1 Align the ferrite bead with the head of the network cable and move the ferrite bead along the cable until there is a gap of 1.0 mm +/- 0.3 mm between the bead and the connector.

Step 2 Place the cable into the ferrite bead channel and loop the cable around the ferrite bead so that the cable exits the bottom of the bead.

Step 3 Hold the cable in the ferrite bead channel and close the ferrite bead. Snap the latch closed.

General IP Phone Information

Power-Saving Mode

Your conference station supports the Cisco EnergyWise (EW) program. Your system administrator sets up sleep (power down) and wake (power up) times for your conference station to save energy. In the sleep state, the LED for the sound base Mute button becomes solid gray, and the DCU powers off.

Ten minutes before the scheduled sleep time, you receive a message that the conference station will power down at a specific time. If the power-saving mode is enabled by your system administrator, you hear your ringtone play. The ringtone plays according to the following schedule:

- At 10 minutes before power down, the ringtone plays four times
- At 7 minutes before power down, the ringtone plays four times
- At 4 minutes before power down, the ringtone plays four times
- At 30 seconds before power down, the ringtone plays 15 times or until the phone powers down

If your conference station is inactive (idle) at the sleep time, it powers down. If your conference station is active (for example, on a call), it waits until it has been inactive for a period of time before removing power.

After the conference station goes to sleep, press the Mute button on the sound base to wake the conference station up. You cannot wake the conference station using the DCU. After the conference station is inactive for a period of time, it goes back to sleep. At the set wake time, the conference station powers up.

Wake and sleep times are also linked to the configured days that you normally work. If your requirements change (for example, your work hours or work days change), contact your system administrator to have your conference station reconfigured.

For more information about EnergyWise and your conference station, contact your system administrator.

Feature Availability

Depending on the conference station system configuration, features included in this guide might not be available to you or might work differently on the conference station. For information, contact your system administrator.

CHAPTER 3

Applications

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- Call History, page 19
- Preferences, page 22
- Linked Mode, page 23
- IP Phone Information, page 24
- Administrator Settings, page 25

Services

Conference Station services can include special telephony features, network data, and web-based information. The services available for your phone depend on the system configuration and the services you are subscribed to. For more information, contact your system administrator.

Access Services

Procedure

Step 1 Press **Apps**.

Step 2 Use the Navigation bar and Select button to scroll and select the service you want to access.

Step 3 Press the **Select** softkey, or press **Exit** to return to the Applications screen.

Call History

Call History allows you to view information about calls made to and from the conference station. A call state icon is displayed to indicate the type of call:

- Received
- Placed
- Missed

The history stores a maximum of 50 records for each history type.

The caller ID displays with the call icon. If the caller ID is unavailable, then the phone number displays. If the phone number is unavailable, then the message `Unknown` displays. All Call History items are integrated into one list and are ordered by time (latest to oldest).

You can dial a number directly from the Call History list.

If set up by your administrator, the Call History logs the received and placed calls from shared lines. To view the call history logs, open the Call History screen.

View Call History

Procedure

Step 1 Press **Apps**.

Step 2 Use the Navigation bar and Select button to scroll and select **Call History**.

View Call Record Details

Procedure

Step 1 Press **Apps**.

Step 2 Use the Navigation bar and Select button to scroll and select **Call History**.

Step 3 Select a call record and press **Details**.
If the **Details** softkey is not visible on the screen, press **More** first.

Step 4 Press ↩ to return to the Call History list.

Dial from Call History

Procedure

Step 1 Press **Apps**.
Step 2 Use the Navigation bar and Select button to scroll and select **Call History**.
Step 3 From the Call History list, select the call you want to dial and press **Call**.

Edit Phone Number

Procedure

Step 1 Press **Apps**.
Step 2 Use the Navigation bar and Select button to scroll and select **Call History**.
Step 3 Select the call you want to edit.
Step 4 Press **Edit Dial**
Step 5 Press >> to move the cursor to the right and press << to move the cursor to the left.

- Press the Up arrow of the Navigation bar to place the cursor before the left-most character.
- Press the Down arrow of the navigation bar to place the cursor after the right-most character.

Step 6 Press to delete numbers.
Step 7 Press **Dial** to dial the edited number.
Step 8 Press to return to the Call History list.

Clear Call History

Procedure

Step 1 Press **Apps**.
Step 2 Use the Navigation bar and Select button to scroll and select **Call History**.
Step 3 Press **Clear**.
Step 4 Press **Delete** to delete the Call History list, or press **Cancel** to go back to the Call History screen.

Delete Call Record

Procedure

Step 1 Press **Apps**.
Step 2 Use the Navigation bar and Select button to scroll and select **Call History**.
Step 3 Select the call you want to delete.
Step 4 Press **Del Call**.
Step 5 Press **Delete** to delete the call, or press **Cancel** to go back to the Call History screen.

Preferences

Ringtones

You can customize how the conference station indicates an incoming call and a new voice mail message. You can also adjust the ringer volume. In addition to the default ringtone, you can customize your phone to have up to six distinctive ring tones. Ring tone preferences can be set directly on your phone or from your User Options web page.

Related Topics

User Options Web Pages, on page 65

Change Ringtone

Procedure

Step 1 Press **Apps**.
Step 2 Use the Navigation bar and Select button to scroll and select **Preferences**.
Step 3 Select **Ringtone**.
Step 4 Select a ringtone.
Step 5 Press **Play** to play the selected ringtone.
Step 6 Press **Set** to select the ringtone.
Step 7 Press **Apply** to confirm your selection, or press **Cancel** to go back to the Ringtone screen.
Step 8 Press ↶ to return to the Preferences screen.

Contrast

You can adjust the contrast level of the LCD screen. The default contrast level is 50%.

Adjust Contrast

Procedure

Step 1 Press **Apps**.

Step 2 Use the Navigation bar and Select button to scroll and select **Preferences**.

Step 3 Select **Contrast**.

Step 4 To increase contrast, press the up arrow on the Navigation bar. To decrease contrast, press the down arrow.

Step 5 Press **Save** to set the contrast level, or press **Cancel** to exit.

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 9: 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 23

IP Phone Information

Phone Information allows you to view the following model information for your phone:

- Model Number
- IP Address
- MAC Address
- Active Load
- Inactive Load
- Last Upgrade

- Active Server
- Stand-by Server
- Mic 1 status
- Mic 2 status
- Wireless Mic 1 RF ID
- Wireless Mic 2 RF ID
- System RF ID
- Linked Mode status
- Backlight On Time
- Backlight On Duration
- Backlight Idle Timeout
- Days Backlight Not Active

To access information about your phone press **Apps** and scroll to Phone Information.

Administrator Settings

For information on accessing and changing the Administrator Settings, contact your system administrator.

CHAPTER 4

Contacts

- Contacts Overview, page 27
- Corporate Directory, page 27
- Personal Directory, page 29

Contacts Overview

Depending on configuration, the conference station provides corporate and personal contacts directories:

- Corporate Contacts: Supports a global corporate directory that you can access on the conference station. Your system administrator sets up and maintains this feature.
- Personal Contacts: If available, supports a personal address book. You can configure and access entries from your conference station and from Cisco Unified Communications Manager User Options web pages.

Corporate Directory

You can use a corporate directory to place calls to coworkers. You can dial a call from the corporate directory even if you are on another call.

The system administrator sets up and maintains the directory.

Tips

Use the numbers on your keypad to enter characters on the conference station screen and use the Navigation bar to move between input fields.

Dial Contact from Search

Procedure

Step 1 Press **Contacts**.

Step 2 Use the Navigation bar and Select button to scroll and select **Corporate Directory**.

Step 3 Select any of these criteria to search for a coworker:

- First name
- Last name
- Number

Step 4 Enter the search criteria information, press **Search**, and select a contact.

Step 5 To dial a contact, perform one of these actions:

- Press **Dial**.
- Press the contact label number.
- Press the **Call** key on the DCU.

Dial Contact from Search During an Active Call

Procedure

Step 1 Press **Contacts**.

Step 2 Use the Navigation bar and Select button to scroll and select **Corporate Directory**.

Step 3 Select any of these criteria to search for a coworker:

- First name
- Last name
- Number

Step 4 Enter the information, then select **Search** > **Dial**.

Step 5 Perform one of these actions to handle the original call:

- Dial the second call to automatically put the first call on hold.
- Press **Transfer** to transfer the first party to the second and drop yourself from the call. To complete the action, press **Transfer** again after dialing.

- Press **Conference** to create a conference call with all parties, including you. To complete the action, press **Conference** again after dialing.
- Press **End Call** to disconnect the first call and dial the second.

Personal Directory

The personal directory, or address book (PAB), contains a list of your personal contacts. You can assign and use fast-dial codes to PAB entries for quick dialing.

The PAB supports a maximum of 99 entries. For each entry you can store:

- Contact name
- Phone number
- Speed dial number

There are two ways to set up and maintain your personal directory:

- Directly on the conference station
- Using your User Options web pages

The system administrator provides the user ID and the PIN that you use to log into the personal directory.

Tips

- Personal Directory automatically logs you out after a certain amount of time. This time limit can vary. For more information, contact your system administrator.
- Use the numbers on the keypad to enter characters on the conference station screen.
- Use the Navigation bar button on the conference station to move between input fields.

The conference station may be set up for international call logging, which is indicated by a plus (+) symbol on the call logs, redial, or call directory entries. For more information, contact your system administrator.

Related Topics

User Options Web Pages, on page 65

Personal Directory Options

The following sections describe the personal directory options.

Sign In and Out of Personal Directory

The conference station automatically logs you out of the PAB after a certain amount of time. This time limit can vary. For more information, contact your system administrator.

Procedure

Step 1 Press **Contacts**.
Step 2 Use the Navigation pad and Select button to scroll and select **Personal Directory**.
Step 3 Enter the user ID and PIN provided by your system administrator and press **Submit**.
Step 4 To logout, select **Log Out**, press **Select**, and then press **OK**.

Add Personal Directory Entry

Procedure

Step 1 Press **Contacts**.
Step 2 Sign in to Personal Directory.
Step 3 Select **Personal Address Book**.
The Search for an entry screen displays.
Step 4 Press **Submit**.
Step 5 Press **New**. You may need to press **More** first.
Step 6 Enter the Name information.
Step 7 Press **Phones** and enter the phone numbers.
Make sure you include any required access codes, such as 9 or 1.
Step 8 Press **Submit** to add the entry to your personal directory.

Related Topics

Sign In and Out of Personal Directory, on page 29
Assign Fast-Dial Code to PAB from the Phone, on page 32

Search for Entry in Personal Directory

Procedure

Step 1 Press **Contacts**.
Step 2 Sign in to Personal Directory.
Step 3 Select **Personal Address Book**.
Step 4 Select the search criteria and press **Submit**.

Related Topics

Sign In and Out of Personal Directory, on page 29
Dial Number from Personal Directory, on page 31

Dial Number from Personal Directory

Procedure

Step 1 Press **Contacts**.
Step 2 Sign in to Personal Directory.
Step 3 Select **Personal Address Book** and search for an entry.
Step 4 Select the personal address book entry that you want to dial.
Step 5 Press **Select**.
Step 6 Press **Dial**.

Delete Personal Directory Entry

Procedure

Step 1 Press **Contacts**.
Step 2 Sign in to Personal Directory.
Step 3 Select **Personal Address Book** and search for an entry.
Step 4 Press **Select**.
Step 5 Press **Edit**.
Step 6 Press **Delete**.
Step 7 Press **OK** to confirm the deletion.

Edit Personal Directory Entry

Procedure

Step 1 Press **Contacts**.
Step 2 Sign in to Personal Directory.
Step 3 Select **Personal Address Book** and search for an entry.
Step 4 Press **Select**.
Step 5 Press **Edit**.
Step 6 Modify the entry information.
Step 7 Press **Phones** to modify a phone number.
Step 8 Press **Update**.

Fast-Dial Codes with Personal Directory

Fast-Dial codes enable you to place calls quickly. You can assign a fast-dial code to an existing PAB entry, or you can assign a code directly to a phone number. Fast-Dial codes that are assigned to numbers rather than to PAB entries are labeled “raw” in the User Options web page. The following sections describe using the fast-dial codes with Personal Directory.

Assign Fast-Dial Code to PAB from the Phone

Procedure

Step 1 Sign in to Personal Directory.
Step 2 Use the Navigation bar and Select button to scroll and select an option.
Step 3 Enter the name information and press **Submit**.
Step 4 Select the name, then press **Select**.
Step 5 Press **FastDial**.
Step 6 Use the Navigation bar combined with **Next** and **Previous** to scroll and select a code.
Step 7 Press **Select** for a new fast-dial code that you want to assign to the number.

Assign Fast-Dial Code to PAB

Procedure

Step 1 Sign in to your User Options web page.
Step 2 Select **User Options** > **Fast Dials**.
Step 3 Select **Add New**.
Step 4 Use the Search Options area to find the appropriate PAB entry
Step 5 Select a phone number in the Search Results area.
Step 6 Change the Fast-Dial code, if desired.
Step 7 Select **Save**.

Place Call Using Fast-Dial Code

Procedure

Step 1 Sign in to Personal Directory.
Step 2 Select **Personal Fast Dials**.
Step 3 Scroll to the fast dial code that you want to dial.
Step 4 Press **Dial**.

Delete Fast-Dial Code

Procedure

Step 1 Sign in to Personal Directory.
Step 2 Select **Personal Fast Dials** and search for a fast-dial code.
Step 3 Select the fast-dial code to delete.
Step 4 Press **Remove**.

CHAPTER 5

Messages

- Access Voicemail, page 35
- Voice Message Identification, page 35
- Access Voice Messages, page 35

Access Voicemail

Procedure

Step 1 Press **Msgs**.

Step 2 Follow the voice prompts.

Voice Message Identification

If you have a voice message, the Call button on the DCU is red and the phone screen displays the New Voicemail status message. You can also configure an audible message indicator using your User Options web pages. If you have configured an audible message indicator, listen for a stutter tone when you try to place a call.

Access Voice Messages

Procedure

Press **Msgs** to listen to voice messages.
You may need to press **More** first.

CHAPTER 6

Calling Features

- Softkey Feature Map, page 38
- Answer, page 41
- Call Back, page 42
- Call Forward, page 42
- Call Park, page 43
- Call Pickup, page 43
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- Cisco Extension Mobility, page 50
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- Hold, page 51
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Softkey Feature Map

Depending upon how your system administrator sets up your phone, not all features may be available to you, some features may be accessed from a different softkey, or additional softkey features are available.

The following table provides a guide to some of the softkey features that are commonly available for various call states. Also shown is the function of the Call button for the matching state.

Table 10: Feature Map

Call State	Softkey 1	Softkey 2	Softkey 3	Softkey 4	Call Button
Idle	Redial	New Call	Apps	More1	Off hook
	Contacts	Msgs	Fwd All	More2	
Ring in	Answer	Divert			Answer
On hook (pre-dial)	Cancel	Call	Speed Dial	X	Dial
Off hook	Redial	Cancel	Apps	More1	On hook
	Contacts	Msgs	Calls	More2	
Ring out	Cancel	Callback			On hook
Connected	Hold	End Call	Apps	More1	On hook
	Contacts	Transfer	Conf	More2	
	ConfList	Park		Top	
On hold	Resume	New Call	Apps	More1	Resume
	Contacts	Msgs	Divert	Top	
Connected (multiple calls)	Hold	End Call	Apps	More1	On hook
	Contacts	Swap or Calls	Transfer	More2	
	Conf	ConfList	Park	Top	

Note The ConfList softkey displays only if you are in a conference.

Note The Calls softkey displays when more than two calls are connected. Pressing Calls loads a call list.

Survivable Remote Site Telephony Overview

If communication between the conference station and the Cisco Unified Communications Manager is interrupted, you receive an alert message on your phone. If you are on an active call, the call remains established, and you enter a failover situation. The Survivable Remote Site Telephony (SRST) feature handles this failover.

While in failover, not all the features of conference station are available. The following table describes typical features and feature availability. For more information about feature availability during failover, contact your system administrator.

When the conference station loses connectivity, it may display a message similar to this: `SRST. Some features unavailable.`

Table 11: Feature Support During Failover

Feature	Supported	Notes
New Call	Yes	
End Call	Yes	
Redial	Yes	
Answer	Yes	
Hold	Yes	
Resume	Yes	
Call Forward	No	This also applies to other forwarding features such as Call Forward All, Call Forward Busy, and Call Forward No Answer.
Conference	Yes	
Conference to Active Calls (Join)	No	The Active Calls softkey is not displayed.
Conference List	No	

Feature	Supported	Notes
Transfer	Yes	Consult only.
Transfer to Active Calls (Direct Transfer)	No	
Auto Answer	Yes	
Call Waiting	Yes	
Caller ID	Yes	
Audible Message Waiting Indicator	Yes	
Unified Session Presentation	Yes	Conference is the only feature supported.
Voicemail	Yes	Your voicemail will not be synchronized with other users in the Cisco Unified Communications Manager cluster.
Speed Dial	Yes	
To Voicemail (Divert)	No	The Divert softkey is not displayed.
Park Monitoring	No	The Park softkey is not displayed.
Barge	No	You see the message `That feature is not currently available.`
Enhanced Message Waiting Indication	No	Message count badges do not appear on the phone screen. Only the Message Waiting icon displays.
Directed Call Park	No	The softkey is not displayed.
Hold Reversion	No	Calls remain on hold indefinitely.
Remote Hold	No	Calls appear as Local Hold calls.
Meet Me	No	The Meet Me softkey is not displayed.
PickUp	No	The softkey causes no action.
Group PickUp	No	The softkey causes no action.

Feature	Supported	Notes
Other PickUp	No	The softkey causes no action.
Malicious Call ID	No	The softkey causes no action.
QRT	No	The softkey causes no action.
Mobility	No	The softkey causes no action.
Privacy	No	The softkey causes no action.
Call Back	No	The Call Back softkey is not displayed.

Answer

Answer allows you to answer the oldest call that is available on the conference station, including Hold Reversion and Park Reversion calls that are in an alerting state. Incoming calls always have priority over Held or Park Reversion calls.

When you get a call, you see a notification on the conference station screen. The call notification remains visible for a preset amount of time. If there are multiple, simultaneous incoming calls, an incoming call list window is displayed, and you can select which call to answer.

To answer a call, press Answer or press the Call key.

If an incoming call has been call forwarded from another phone or conference station, you may see additional information to identify that the call has been forwarded. Your system administrator controls the amount of additional information displayed. Additional information can identify the person who forwarded the call to you and the caller information.

When you receive a call, the phone number that displays on the screen contains the string of digits that you can dial to contact the caller. The digit string can contain the following digits, if required:

- Code to obtain an outside line (for example, if you have to dial 9)
- Long-distance code
- Area code
- City code
- Telephone number

The conference station saves the complete digit string in the call history and you can save the number in your Personal Address Book.

Call Back

Call Back allows you to receive an audio and a visual notification on your conference station when a busy or unavailable party becomes available.

For more information, contact your system administrator.

Set Up Call Back Notification

Procedure

Step 1 Press **Call back** while listening to the busy tone or ring sound.
A confirmation screen displays on the phone.

Step 2 Press **Exit** to exit the confirmation screen.
Your phone alerts you when the line is free.

Step 3 Press **Call** to place the call again.

Call Forward

Call Forward allows you to forward incoming calls from the conference station to another number.

There are two types of call forwarding features that the system administrator can set up for the conference station:

- Unconditional call forwarding (Call Forward All): Applies to all calls that you receive.
- Conditional call forwarding (Call Forward No Answer, Call Forward Busy, Call Forward No Coverage): Applies to certain calls that you receive.

If configured, you can set up Call Forward All from the conference station. Call Forward All can also be accessed remotely from your User Options web pages. Conditional call forwarding rules can only be accessed from your User Options web pages.

When forwarding calls from your conference station:

- Enter the call forward target number exactly as you would dial it from the conference station. For example, enter an access code or the area code, if necessary.
- Call forwarding is line specific. If a call reaches you on a line where call forwarding is not enabled, the call rings as usual.
- For more details about the following configurable call forward options, contact your system administrator:
  - Allow calls placed from the call forward target number to the conference station to ring through, rather than be forwarded.
  - Prevent you from creating a call forward loop or exceeding the maximum number of links in a call forwarding chain.

Forward Calls

Procedure

Step 1 Press **Fwd All**.

Step 2 Enter the target phone number.
Depending on how your voicemail system is set up, you may be able to press **Msgs** to forward all calls to voicemail.
A visual confirmation displays on the screen while call forwarding is active.

Step 3 To cancel call forwarding, press **Fwd OFF**.

Step 4 To forward calls remotely or to set conditions on fall forwarding, go to your User Options Web pages.

Call Park

Call Park allows you to use the conference station to park (temporarily store) a call. The parked call can be retrieved from another phone in the Cisco Unified Communications Manager system, such as a phone at a coworker's desk or in a conference room.

You retrieve a parked call by entering the parking number for the call into another phone in the Cisco Unified Communications Manager system.

Your system administrator sets up the parking number for the conference station.

Park Call

Procedure

Step 1 During a call, press the **Park**.
The conference station screen displays the call park number where the system stored your call.

Step 2 Note the call park number displayed on the screen. This number is used to retrieve the call.
You have a limited time to retrieve a parked call before it reverts to ringing at the original number.

Step 3 From any other Cisco Unified IP Phone or conference station in your network, enter the call park number to retrieve the call.

Call Pickup

Call Pickup allows you to answer a call that is ringing on a coworker's phone by redirecting the call to your phone.

You might use Call Pickup if you share call-handling tasks with coworkers.

The ways to pick up a call are:

- Pickup: Allows you to answer a call that is ringing on another phone within your call pickup group.

  If multiple calls are available for pick up, the conference station picks up the oldest call first.
- Group Pickup: Allows you to answer a call on a phone that is outside your call pickup group by:
  - Using a group pickup number that is provided by your system administrator.
  - Dialing the number of the ringing phone.
- Other Pickup: Allows you to answer a call that is ringing on another phone within in your call pickup group or in an associated call pickup group.

Your system administrator assigns you to a call pickup group and sets the call pickup softkeys.

Answer Call Using Pickup

Procedure

Step 1 Press **Pickup** to transfer a ringing call within your pickup group to the conference station.

Step 2 If the call rings, press **Answer** to connect to the call.

Answer Call Using Group Pickup and Group Pickup Number

Procedure

Step 1 Press **Group Pickup** to answer a call on a phone outside your pickup group.

Step 2 Enter the group pickup number.

Step 3 If the call rings, press **Answer** to connect to the call.

Answer Call Using Group Pickup and Phone Number

Procedure

Step 1 Press **Group Pickup**.

Step 2 Enter the number of the phone line with the call that you want to pick up.
For example, if the call is ringing on line 12345, enter 12345.

Step 3 If the call rings, press **Answer** to connect.

Answer Call Using Other Pickup

Procedure

Step 1 Press **OPickup** to transfer a call in your pickup group or in an associated group to the conference station.

Step 2 If the call rings, press **Answer** to connect.

cBarge

The cBarge feature allows you to add yourself to a call on a shared line and create a standard (ad hoc) conference.

Join Conference on Shared Line

Procedure

Press **cBarge**.
You may need to press **More** first.

Conference

Conference allows you to talk simultaneously with multiple parties.

When you are talking on a call, use Conference to dial another party and add them to the call.

Before completing a conference procedure, you can press Cancel to cancel the procedure.

As the conference host, you can remove individual participants from the conference. You can also view a list of participants.

The conference ends when all the participants hang up.

Add Third Party to Conference

Before You Begin

Before you can add a party to the conference, you must be on an active call and not on hold.

Procedure

Step 1 Press **Conf**.

Step 2 Enter the phone number, fast-dial code, or speed-dial number for the party you want to add.
If you have several held calls, you can press **Calls** to display a caller list and add a caller to the conference.

Step 3 After the new party answers, press **Conf**.
The conference begins.

Step 4 (Optional) Repeat these steps to add more parties, if desired.

View Conference Participants

Procedure

While in a conference, press **ConfList** to view a list of the last 16 participants who have joined the conference. The maximum number of participants that can be displayed is 16. If there are more than 16 participants, only the most recent 16 participants to join display.

Remove Conference Participants

Procedure

Step 1 While in a conference, press **ConfList** to view a list of participants.

Step 2 Highlight the participant that you want to remove and then press **Remove**.

Note Only the most recent 16 conference participants
display.

Divert

Divert allows you to send an active or ringing call to your voicemail system or to a predetermined phone number. Your system administrator configures this feature and sets the receiving phone number.

Divert Call

Procedure

Press **Divert** to send an active call, an incoming call, or a held call to either your voicemail system or to a predetermined phone number set up by your system administrator.

Do Not Disturb

Do Not Disturb (DND) allows you to turn off notification of incoming calls. The ringer, as well as audible and visual notifications, can be turned off.

Depending on how your administrator has configured this feature, incoming calls are either immediately rejected, or the caller information displays on screen.

The system administrator configures the DND softkey, but you can change your DND options from your User Options web pages.

DND interacts with other types of calls:

- If both DND and Call Forward All are enabled, calls are forwarded without any visual or audible confirmation.
- DND does not affect priority calls.

Turn DND On and Off

Procedure

Step 1 Press **DND** to turn on DND.
Visual confirmation displays briefly.

Step 2 Press **DND** again to turn off DND.
Visual confirmation displays briefly.

Enhanced Room Coverage

Optional microphone extension kits provide enhanced room coverage that can be further expanded by linking two units together in Linked Mode. This feature allows you to use the conference station in a larger room, or to enhance the audio and voice experience for larger groups of in-person attendees.

When two conference station base units are linked together to expand the audio coverage area, one conference station acts as the primary device and the other sound base is the dependant or secondary device. In Linked Mode, the primary base station supports one or two wireless microphones, or it supports one wired microphone. The secondary unit supports only one wired microphone; a wireless microphone cannot be connected to a

secondary Sound Base. You cannot mix microphone types on the devices. Consult the following tables for allowable configurations.

Table 12: Deployment Configurations for Enhanced Room Coverage with a Single Phone

	Wired Extension Microphone	Wireless Extension Microphone
Number and type of connected microphones	1 or 2	--
	--	1 or 2

Table 13: Deployment Configurations for Enhanced Room Coverage in Linked Mode

	Primary Sound Base	Secondary Sound Base
Wired Extension Microphone	--	--
Wired Extension Microphone	--	1
Wired Extension Microphone	1	--
Wired Extension Microphone	1	1
Wireless Extension Microphone	--	--
Wireless Extension Microphone	1 or 2	--

Voice, dial tone, ringer, and base LED features synchronize between the two devices in Linked Mode.

Note Use a daisy cable to connect two sound base units in Linked Mode.

Related Topics

Linked Mode, on page 23

Pair Wireless Microphone

Before You Begin

The microphone must be in the off state before you can pair it to the conference station. A microphone is off if the microphone's LED is off. To turn off the wireless microphone, hold down the microphone button until the microphone LED turns solid red, then release.

Procedure

Step 1 Choose **Apps** > **Admin Settings** > **Wireless Microphones**.

Step 2 Select either Wireless Microphone 1 or Wireless Microphone 2.
If the selected channel is available, a `Pair microphone 1?` or `Pair microphone 2?` prompt displays, and the **Pair** softkey displays.

If a microphone is already linked to a particular channel, pairing cannot be initiated on the selected channel and the dialog shows that the microphone is linked.

Step 3 Press **Pair**.
If the channel is ready to pair, the pairing process begins and a text message displays.

Step 4 Put the microphone that corresponds to the selected channel in pairing mode by pressing the microphone's Mute button until the LED lights solid red.
If pairing succeeds, the screen reverts to the Wireless Microphones Menu, and the message `Mic X Paired Successfully!` displays.

If pairing times out or fails, the status is updated and you can cancel or retry.

Step 5 Press **Cancel** to revert to the Wireless Microphones Menu.

Step 6 Press **Retry** to start the pairing process again.

Related Topics

Unpair Wireless Microphone, on page 49

Unpair Wireless Microphone

If you need to connect a wired microphone to the conference station, any wireless microphones must be unpaired first. You can also use this procedure to unpair a microphone that is no longer in use.

Note This option is not available if the microphone is connected. To enable the unpair command, place the wireless microphone in its charger or turn it off.

Procedure

Step 1 Choose **Applications** > **Admin Settings** > **Wireless Microphones**.

Step 2 Select either Wireless Microphone 1 or Wireless Microphone 2.
If the selected channel is paired, the **Unpair** softkey displays.

Step 3 Press **Unpair**.
A verification prompt with the options to Cancel or **Unpair** displays.

Step 4 Press **Unpair** to continue to unpair the microphone.

The microphone channel's registration information in the base deletes. If you view the microphone channel's status in phone info menu, the status value and RFID are empty.

Step 5 Press return to revert to the wireless microphones menu and stop the process.

Related Topics

Pair Wireless Microphone, on page 48

Cisco Extension Mobility

Cisco Extension Mobility allows you to temporarily configure a Cisco Unified IP Phone or Conference Station to use as your own. After you log in to Extension Mobility, the new device adopts your user profile, features, established services, and web-based settings. Your system administrator must configure Extension Mobility for you.

The Cisco Extension Mobility ChangePIN feature allows you to change your PIN from your Cisco Unified IP Phone.

Note

- Extension Mobility automatically logs you out after a certain amount of time. Your system administrator establishes this time limit.
- Changes that you make to your Extension Mobility profile from your User Options web pages take effect immediately if you are logged in to Extension Mobility on the physical device; otherwise, changes take effect the next time you log in.
- Changes that you make to a phone or conference station from your User Options web pages take effect immediately if you are logged out of Extension Mobility; otherwise, changes take effect after you log out.
- Local settings controlled by the phone are not maintained in your Extension Mobility profile.

Enable Extension Mobility

Procedure

Step 1 Press **Apps**.

Step 2 Select **EM Service**.

Step 3 Enter your user ID and PIN.
The user ID and PIN are provided by your system administrator

Step 4 If prompted, select a device profile.

Step 5 To sign out, press **Apps**.

Step 6 Select **Services**.

Step 7 Select **EM Service**.

Step 8 At the prompt, press **Yes**.

Fast Dial

Fast Dial allows you dial a phone number from the Fast Dial service on your phone. Before you can use Fast Dial on your phone, you must set up Fast Dial on your User Options web pages.

Related Topics

User Options Web Pages, on page 65

Place Call Using Fast Dial

Before you can use Fast Dial on the conference station, it must be set up on your User Options web pages.

Procedure

Step 1 Press **Contacts**.

Step 2 Sign in to Personal Directory.

Step 3 Select **Personal Fast Dials**.

Step 4 Select a fast-dial code and then press **Call**.

Hold

Hold allows you to put an active call into a held state. Your phone allows one active call at a time; other calls are put on hold.

Hold Active Call

Procedure

Step 1 To put an active call on hold, press **Hold**.

The Hold icon displays as a status icon.

Step 2 If there is only one call on hold and you are not on an active call, press **Resume**.

Step 3 If you are already on an active call, press **Swap**.
The holding call becomes active, and the active call is placed on hold.

Switch Between Active and Multiple Holding Calls

Procedure

Step 1 If you are on an active call and there are multiple calls on hold, the **Calls** softkey becomes available and a call list of holding calls displays on the screen.

Step 2 Use the Navigation bar to highlight the call you want to make active, press **Resume**.
The current active call is placed on hold and the selected call is now active.

Hold Active Call and Answer New Incoming Call

Procedure

To place an active call on hold and answer an incoming call, press**Answer**.

Hold Reversion

Hold Reversion provides a notification that a call is left on hold. Hold reversion notifications are similar to new call notifications.

Depending on the configuration of the conference station, you may see one or more of the following:

- An animated icon that appears as an incoming call for two seconds and then as a hold icon for two more seconds.
- A single ring that repeats at regular intervals.
- The Call button on the DCU flashes green.
- The LEDs on the sound base flash green.

Respond to Hold Reversion Notification

Procedure

Press **Answer**

Meet Me Conference

If enabled by your system administrator, you can call a predetermined number at a scheduled time to host or join a Meet Me conference.

The Meet Me conference begins when the host connects. Participants who call the conference before the host has joined hear a busy tone and must dial again.

Note The conference ends when all participants hang up; the conference does not automatically end when the host disconnects.

Host Meet Me Conference

Procedure

Step 1 Obtain a Meet Me phone number from your system administrator.

Step 2 Distribute the Meet Me phone number to participants.

Step 3 When you are ready to start the meeting, press **MeetMe**.
You may need to press **More** first.

Step 4 Dial the Meet Me phone number.

Join Meet Me Conference

Procedure

Step 1 Dial the Meet Me phone number provided by the conference host.

Step 2 If you hear a busy tone, the host has not yet joined the conference. In this case, disconnect and try your call again.

Mobile Connect

Mobile Connect allows you to use your mobile phone to handle calls associated with the conference station phone number.

To set up Mobile Connect, use the User Options web pages to set up remote destinations and create access lists to allow or block calls from specific phone numbers from being passed to the remote destinations.

When you enable Mobile Connect:

- The conference station and remote destinations receive calls simultaneously.
- When you answer the call on the conference station, the remote destinations stop ringing, disconnect, and display a missed call message.
- When you answer the call on one remote destination, the other remote destinations and the conference station stop ringing, disconnect, and a missed call message displays on the other remote destinations.

Enable Mobile Connect

Procedure

Step 1 Press **More**.

Step 2 Press **Mobility** to display the current remote destination status (Enabled or Disabled).

Step 3 Press **Select** to change the status.

Step 4 Press **Exit**.

Switch IP Phone Call to Mobile Phone

Procedure

Step 1 Press **More**.

Step 2 Press **Mobility**.

Step 3 Select **To mobile**.

Step 4 Answer the in-progress call on your mobile phone.
You cannot use the conference station for any other calls while this call is still in progress.

The Call button LED lights solid red, and the calling party number displays on the phone.

Switch Mobile Call to IP Phone

Procedure

Step 1 Press **Call** on the conference station.

Step 2 Hang up the call on your mobile phone to disconnect the mobile phone, but not the call.

Step 3 Press **Resume** on the conference station within 5 to 10 seconds and start talking on the conference station.

Monitoring and Recording

The Monitoring and Recording feature allows you to monitor and record calls. Your system administrator enables this feature, which can be set up for automatic recording of all calls or recording of calls on an individual call basis.

You can start or stop a recording by pressing the Record softkey on your phone.

Users might receive audible alerts during call monitoring and recording. By default, the person who monitors and records the call does not receive an audible alert.

Secure monitoring and recording is also available. For details on this aspect of the feature, contact your system administrator.

Multiple Calls per Line

The conference station has a single line and supports a maximum of six calls. Unless you are in a conference, only one call can be connected at any time; other calls are automatically placed on hold.

Multiple Incoming Calls

If there is a second incoming call on the line, while the first call rings, a navigable incoming call list window displays on the screen. The call list window updates automatically if there are additional incoming calls, or if an incoming call is cancelled.

Call Ended on Line with Multiple Calls

If there are multiple calls on the line when a call completes, the next call in the call list gains focus.

Outbound Call Maximum

The conference station supports a maximum of 6 outbound calls.

Answer Second Call on Same Line

Procedure

To answer a second call on your conference station line, press **Answer**.

Any active call is placed on hold, and the second call is answered.

Switch Between Calls on Same Line

Procedure

Step 1 Press **Swap** to switch between two calls on the same line.

Step 2 If there are more than two calls on the line, select a call from the call list and press **Resume**.

Create Conference with Two Calls on Same Line

Procedure

Step 1 With two connected calls on the same line, select a call to make it the active call.
The second call is put on hold.

Step 2 Press **Conference**.

Step 3 Press **Calls** to view the call list and select the caller to add to the conference.

Step 4 Wait for the call to connect.

Step 5 Press **Conference** to add the participant to your call.
The conference begins.

Step 6 (Optional) Repeat to add additional participants.

Transfer Two Calls on Same Line

Before You Begin

You must be on an active call to transfer calls.

Procedure

Step 1 Press **Transfer**.

Step 2 Enter phone number or press a speed-dial button for the transfer destination.
When you have reached the maximum number of calls for your line, pressing **Transfer** allows you to select the calls from a list of calls on the line.

Step 3 Wait for the recipient to answer.

Step 4 Press **Transfer** again.

Shared Line

If the conference station is registered on a shared line you can handle multiple calls in the following ways:

- If there are two or more remote calls on the shared line, the caller ID field on the conference station indicates the number of remote calls on the line, and the state of the call. Press Calls to display the call list window.
- If at least one call on the shared line is on hold, the Call button LED flashes red. Press Resume to make the call active, or use the Navigation bar to choose a call to resume from the call list. For example, if a remote call on the shared line is placed on hold the Call button on your conference station will flash red.

Mute

Mute allows you to block audio input for the conference station, so that you can hear other parties on the call but they cannot hear you.

The conference station can be muted in two ways:

- Press the Mute button on the Sound Base.
- Press the Mute button on the DCU.

If connected, the optional extension microphones can also be used to mute the conference station. In Linked Mode, the Mute button on the secondary sound base behaves the same as the Mute button on the primary unit.

LED indicators on both the sound base and the DCU indicate the mute status of the conference station:

- Solid, red base LEDs and red base Mute button: muted.
- Solid, red DCU Mute button: muted.

Mute IP Phone Sound Base

Procedure

Step 1 Press **Mute** to turn Mute on.
The **Mute** button is backlit by a solid, red light, and the sound base LEDs light red.

Step 2 Press **Mute** again to turn Mute off.

Mute IP Phone DCU

Procedure

Step 1 Press **Mute** to turn Mute on.
The **Mute** button is backlit by a solid, red light, and a mute icon displays on the screen.

Step 2 Press **Mute** again to turn Mute off.

On-Hook Dialing

On-hook dialing allows you to enter a phone number before getting a dial tone and then press the Call button to complete the call.

Dial Number On-Hook

Procedure

Step 1 Enter or speed dial a phone number.

Step 2 Press **Dial**.

Plus Dialing

Plus Dialing allows you to press and hold the star (*) key for at least 1 second to insert a plus (+) sign as the first digit in a phone number when dialing an international number.

A phone number with the + sign in it can be selected and dialed without the need to add digits for international calls.

Dial International Number

Before You Begin

Before dialing an international call or using Plus Dialing, enter any local access code, such as 8 or 9 if applicable.

Procedure

Step 1 Press and hold star (*) for at least 1 second.
The plus (+) sign displays as the first digit in the phone number.
The corresponding key tone stops to indicate that the * has changed to a + sign.

Step 2 Dial the international number, including the country code.

Privacy

If configured, the Privacy feature allows you to prevent others who share your line from seeing information about your calls. Your system administrator configures this feature.

If the phone that shares your line has Privacy enabled, you can make and receive calls using the shared line as usual.

Enable Privacy on Shared Line

Before You Begin

Before you can use this feature, it must enabled by your system administration.

Procedure

Step 1 Press **More**.

Step 2 Use the Navigation bar and Select button to scroll to the **Private** softkey.

- Press **Private** to enable this feature.
- Press **Private** to disable this feature if it is currently enabled.

Quality Reporting Tool

Your system administrator may temporarily configure the conference station with the Quality Reporting Tool (QRT) to troubleshoot performance problems. Invoking the QRT reports a problem with the current call to the system administrator.

Report IP Phone Problems

Procedure

Step 1 Press **More**.

Step 2 Use the Navigation bar and Select button to locate and select **QRT**.
The information is sent to your system administrator.

Redial

Redial allows you to call the most recently dialed phone number.

Redial Number

Procedure

To redial the last number you called, press **Redial**.

Shared Lines

Shared lines allow you to use one phone number for multiple devices.

A shared line is useful if you have multiple devices and want one phone number, share call-handling tasks with coworkers, or handle calls on behalf of a manager. The other devices that share your line are referred to as remote devices, and a call that is being handled by a device that shares your line is referred to as a remote call.

When a call comes in on the shared line, the conference station rings and your coworker's device rings. Either you or your coworker can answer the call, place the call on hold, or transfer the call.

The Call button on the DCU has an LED that indicates the call state. For example:

- Remote in use: solid red LED.
- Remote hold: pulsing red LED

Your call history shows the status for all calls on the shared line. For example, if a call rings on a shared line and you answer the call, your coworkers who share the line see that the call was answered remotely. Your call history identifies calls that were Placed, Received, or Missed.

Speed Dial

Speed-Dial features allow you to quickly place a call using preconfigured codes to place a call, or by selecting an item from the Call History list. Before you can use speed-dial features on the conference station, you must set up Speed Dial on your User Options web pages. The conference station allows 10 speed-dial entries.

The conference station supports these speed-dial features:

- Speed-dial codes: Allows you to dial a phone number from a code (sometimes referred to as abbreviated dialing or fast dialing).

Related Topics

Speed Dial Setup on Web, on page 70

Place Call Using Speed Dial Codes

Before You Begin

Before you can use speed-dial codes on the conference station, you must set up the codes.

Procedure

To place a call, enter the speed-dial code and press the **Speed Dial**.

Transfer

Transfer allows you to redirect a connected call from the conference station to another number.

Before completing a transfer procedure, you can press **Cancel** to cancel the procedure.

Transfer Call to Another Number

Before You Begin

The call must be active to be transferred.

Procedure

Step 1 Press **Transfer**.

Step 2 Enter the destination number or press a speed-dial button.

Step 3 Wait for the recipient to answer.

Step 4 Press **Transfer** again.
The transfer completes.

Cisco WebDialer

Cisco WebDialer allows you to click-to-dial contacts from the Cisco Unified Communications Manager Directory. Your system administrator sets up this feature for you.

Use WebDialer with Cisco Directory

Procedure

Step 1 Sign in to your User Options web pages.

Step 2 Select **User Options** > **Directory** and search for a coworker.

Step 3 Select the number that you want to dial.

Step 4 If this is your first time using WebDialer, review the preferences on the Make Call window.

Step 5 Select **Dial**.
The call is now placed on your phone.

Step 6 To end a call, select **Hang up** from the Make Call window, or press **End Call** on your phone.

Use WebDialer with Another Online Corporate Directory

Procedure

Step 1 Sign in to a WebDialer-enabled corporate directory and search for coworkers.

Step 2 Select the number that you want to dial.

Step 3 When prompted, enter your user ID and password.

Step 4 If this is your first time using WebDialer, review the preferences on the Make Call window.

Step 5 Select **Dial**.

The call is now placed on your phone line.

Step 6 To end a call, select **Hang up** in the Make Call window or press **End Call** on your conference station.

Change WebDialer Preferences

Procedure

Step 1 Sign in to your User Option web pages.

Step 2 Initiate a call using WebDialer to access the Make Call window.
The Make Call window displays the first time that you use WebDialer (after you select the number that you want to dial).

Step 3 Select one of the following options from the Make Call window:

- Preferred language: Determines the language used for WebDialer settings and prompts.
- Use preferred device: Identifies the Cisco Unified IP Phone (calling device) and directory number (calling line) that you use to place WebDialer calls.

If you have one phone with a single line, the appropriate phone and line are automatically selected.

Step 4 If the phone and line do not select automatically, choose a phone or line.
If you have more than one phone of the same type, the list identifies the phone by device type and MAC address. To display the MAC address on your phone, select **Apps** > **Phone Information**.

Step 5 If you have an Extension Mobility profile, select **Extension Mobility** from the Calling Device drop-down menu in the Make Call window.

Step 6 Ensure that you do not select **Do not display call information** or **Disable Auto Close**.

- Do not display call confirmation: If selected, the WebDialer Make Call window does not display the next time WebDialer is used. Calls will automatically dial after you select a contact from the Cisco directory.
- Disable Auto Close: If selected, the call window does not close automatically after 15 seconds.

CHAPTER 7

User Options Web Pages

- User Options Web Pages Overview, page 65
- Sign In and Out of User Options Web Pages, page 65
- Select Device from User Options Web Page, page 66
- Features and Services Setup on Web, page 66

User Options Web Pages Overview

Your Cisco Unified IP Conference Phone 8831 is a network device that can share information with other network devices in your company, including your personal computer. You can use your computer to log in to your User Options web pages, where you can control features, settings, and services for your conference station.

This section describes how to log in and select a conference station.

Sign In and Out of User Options Web Pages

Before you can access any of your user options, such as Speed Dial or Personal Address Book, you must sign in. When you are finished using the User Options web pages, you must sign out.

In some cases, you can access your User Option web pages without having to sign in. For more information, contact your system administrator.

Procedure

Step 1 Obtain the User Options URL, user ID, and default password from your system administrator.

Step 2 Open a web browser on your computer and enter the URL.

Step 3 If prompted to accept security settings, select **Yes** or **Install Certificate**.

Step 4 Enter your user ID in the **Username** field.

Step 5 Enter your password in the **Password** field.

Step 6 Select **Login**.

The Cisco Unified CM User Options home page displays. From this page you can use User Options to select a device, access User Settings, Directory features, your Personal Address Book, and Fast Dials.

Step 7 To sign out of user options, select **Logout**.

Select Device from User Options Web Page

Procedure

Step 1 Log in to your User Options web pages, and then choose **User Options** > **Device**.
The Device Configuration page appears.

Step 2 If you have multiple conference stations assigned to you, verify that the appropriate conference station is selected. If necessary, choose another conference station from the Name drop-down list.
From the Device Configuration page, you can access conference station-specific options, such as Line Settings, Speed Dials, Phone Services, and Service URL.

Step 3 To return to the Device Configuration page from another page, choose **User Options** > **Device**.

Features and Services Setup on Web

This section describes how to configure features and services from your User Options web pages after logging in.

Related Topics

Sign In and Out of User Options Web Pages, on page 65

Personal Directory on Web

The Personal Directory feature set that you can access on your computer consists of:

- A Personal Address Book (PAB)
- Fast Dials
- Cisco Unified Communications Manager Address Book Synchronizer

You can also access PAB and Fast Dials from the conference station.

Related Topics

Personal Directory, on page 29

Personal Address Book on Web

This section describes how to use your PAB from your User Options web pages.

Related Topics

Sign In and Out of User Options Web Pages, on page 65

Add New Personal Address Book Entry

Procedure

Step 1 Choose **User Options** > **Personal Address Book**.
Step 2 Click **Add New**.
Step 3 Enter information for the entry.

Search Personal Address Book Entry

Procedure

Step 1 Choose **User Options** > **Personal Address Book**.
Step 2 Specify search information, and then click **Find**.

Edit Personal Address Book Entry

Procedure

Step 1 Choose **User Options** > **Personal Address Book**.
Step 2 Search for the PAB entry you want to edit.
Step 3 Click the nickname for the PAB entry you want to edit.
Step 4 Edit the entry, and then click **Save**.

Delete Personal Address Book Entry

Procedure

Step 1 Choose **User Options** > **Personal Address Book**.
Step 2 Search for the PAB entry you want to delete.
Step 3 Check the checkbox next to the PAB entry you want to delete.
Step 4 Click **Delete Selected**.

Fast Dials on Web

This section describes how to assign Fast Dials from your User Options web pages.

Tips

- You can create up to 500 fast-dial and PAB entries.
- When you create a new fast-dial entry without using a PAB entry, the fast-dial entry is labelled “Raw” in your User Options web pages. The entry does not display a configurable text label and does not have a matching entry in your PAB.

Related Topics

Sign In and Out of User Options Web Pages, on page 65

Assign Fast-Dial Code to Personal Directory Entry

Procedure

Step 1 Sign in to Personal Directory.
Step 2 Enter the name information and press **Submit**.
Step 3 Select the name, then press **FastDial**.
Step 4 Select a number and press **Select**.
Step 5 Press **Next**.
Step 6 Press **Assign** for a new fast-dial code that you want to assign to the number.

Assign Fast-Dial Code to Phone Number

Procedure

Step 1 Sign in to your User Options web page.
Step 2 Select **User Options** > **Fast Dials**.
Step 3 Select **Add New**.
Step 4 Change the Fast-Dial code, if desired.
Step 5 Enter a phone number.
Step 6 Select **Save**.

Search for Fast-Dial Entry

Procedure

Step 1 Sign in to your User Options web page.
Step 2 Select **User Options** > **Fast Dials**.
Step 3 Specify search information and select **Find**.

Edit Fast-Dial Phone Number

Procedure

Step 1 Sign in to your User Options web page.
Step 2 Select **User Options** > **Fast Dials**.
Step 3 Search for the Fast-Dial entry that you want to edit.
Step 4 Select a component of the entry.
Step 5 Change the phone number.
Step 6 Select **Save**.

Delete Fast-Dial Personal Address Book Entry

Procedure

Step 1 Sign in to your User Options web page.

Step 2 Search for a fast-dial entry.

Step 3 Select one or more entries.

Step 4 Select **Delete Selected**.

Address Book Synchronization Tool

You can use the Cisco Unified Communications Manager Address Book Synchronizer to synchronize your existing Microsoft Windows Address Book (if applicable) with your PAB. Entries from your Microsoft Windows Address Book become accessible on your conference station and your User Options web pages. Your system administrator can give you access to the synchronizer and provides detailed installation instructions.

Speed Dial Setup on Web

Depending on configuration, the conference station can support several speed-dialing features:

- Speed Dials
- Abbreviated Dials
- Fast Dials

This section describes how to set up speed-dialing features from your User Options web pages.

Related Topics

Speed Dial, on page 61
Sign In and Out of User Options Web Pages, on page 65
Fast Dials on Web, on page 68

Set Up Speed-Dial Codes

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.
Step 2 Select a phone from the Name drop-down menu.
Step 3 Select **Speed Dials**.
Step 4 In the Speed Dial Settings area, enter a number and label for a speed-dial code.
Step 5 Select **Save**.

Set Up Abbreviated Dialing Codes

Procedure

Step 1 Choose **User Options** > **Device**.
Step 2 Choose a phone from the Name drop-down menu.
Step 3 Click **Speed Dials**.
Step 4 Enter a number and label for an Abbreviated Dialing code.
Step 5 Click **Save**.

Phone Services Setup on Web

Conference station services can include special conference station features, network data, and web-based information (such as stock quotes and movie listings). You must first subscribe to a conference station service before accessing it on the conference station.

This section describes how to configure conference station services using your User Options web pages.

Related Topics

Sign In and Out of User Options Web Pages, on page 65

Subscribe to Service

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.
Step 2 Select a phone from the **Name** drop-down menu.
Step 3 Select **Phone Services**.
Step 4 Select **Add New**.
Step 5 Select a service from the drop-down list and select **Next**.
Step 6 (Optional) Change the service label or enter additional service information, if available.
Step 7 Select **Save**.

Search for Services

Procedure

Step 1 On your User Options web page, select a device.
Step 2 Select **Phone Services**.
Step 3 Select **Find**.

Change or End Services

Procedure

Step 1 On your User Options web page, search for services.
Step 2 Select one or more entries.
Step 3 Select **Delete Selected**.

Change Service Name

Procedure

Step 1 On your User Options web page, search for services.

Step 2 Select a service name.

Step 3 Change the information and select **Save**.

Access Service on Phone

Procedure

From the conference station, choose > **Services**.

User Settings on Web

User settings include your password, PIN, and language (locale) settings.

Your PIN and password allow you to access different features and services. For example, use your PIN to log in to Cisco Extension Mobility or Personal Directory on the conference station. Use your password to log in to your User Options web pages and Cisco WebDialer on your personal computer. For more information, contact your system administrator.

This section describes how to control your user settings from your User Options web pages.

Related Topics

Sign In and Out of User Options Web Pages, on page 65

Change Browser Password

Procedure

Step 1 Sign in to your User Options web pages.

Step 2 Select **User Options** > **User Settings**.

Step 3 Enter your Current Password.

Step 4 Enter your New Password.

Step 5 Reenter your new password in the Confirm Password field.

Step 6 Select **Save**.

Change PIN

Procedure

Step 1 Sign in to your User Options web pages.
Step 2 Select **User Options** > **User Settings**.
Step 3 Enter your Current PIN.
Step 4 Enter your New PIN.
Step 5 Reenter your new PIN in the **Confirm PIN** field.
Step 6 Select **Save**.

Change User Options Web Page Language Setting

Procedure

Step 1 Sign in to your User Options web page.
Step 2 Select **User Options** > **User Settings**.
Step 3 In the User Locale area, select an item from the Locale drop-down list.
Step 4 Select **Save**.

Change Phone Display Language

Procedure

Step 1 Sign in to your User Options web page.
Step 2 Select **User Options** > **Device**.
Step 3 Select an item from the User Locale drop-down list.
Step 4 Select **Save**.

Line Settings on Web

Line settings include call forward, voice message indicators, ring types, and line text labels that you set up for your conference station.

Note By default, the ring type and line text label options do not display on your User Options web pages. Contact your system administrator to set up these options so that you can access them.

This section describes the line settings you can update from your User Options web pages.

Related Topics

Sign In and Out of User Options Web Pages, on page 65
Call Forward, on page 42
Preferences, on page 22

Set Up Call Forward per Line

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.

Step 2 Select a phone from the Name drop-down menu.

Step 3 Select **Line Settings**.

Step 4 If you have more than one directory number (line) assigned to your phone, select a line from the Line drop-down menu.

Step 5 In the Incoming Call Forwarding area, select call forwarding settings for various conditions.

Step 6 Select **Save**.

Change Voice Message Indicator Setting per Line

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.

Step 2 Select a phone from the **Name** drop-down menu.

Step 3 Select **Line Settings**.

Step 4 (Optional) If you have more than one directory number (line) assigned to your phone, select a line from the Line drop-down menu.

Step 5 In the Message Waiting Lamp area, choose from various settings.
Typically, the default message waiting setting prompts your phone to display a steady red light from the handset light strip to indicate a new voice message.

Step 6 Select **Save**.

Change Audible Voice Message Indicator Setting per Line

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.

Step 2 Select a phone from the **Name** drop-down menu.

Step 3 Select **Line** Settings.

Step 4 If you have more than one directory number (line) assigned to your phone, select a line from the Line drop-down menu.

Step 5 In the Audible Message Waiting Lamp area, choose from various settings.

Step 6 Select **Save**.

Change Ring Type

When the conference station is idle, an incoming call rings uses the idle ring setting. When the conference station is active, an incoming call rings uses the consecutive ring setting.

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.

Step 2 If you have more than one device, select a device.

Step 3 Select **Line Settings**.

Step 4 In the Ring Settings area, choose a setting to tell the conference station how to indicate an incoming call.

Step 5 Select **Save**.

Update Line Text Label Displayed on Called Party Phone

Procedure

Step 1 On your User Options web page, select **User Options** > **Device**.

Step 2 If you have more than one device, select a device.

Step 3 Select **Line Settings**.

Step 4 In the Line Text Label area, enter the text that will identify your directory number when you call another conference station.
If you do not specify a line text label, your directory number displays on the called party's phone.

Step 5 Select **Save**.

Cisco WebDialer

Cisco WebDialer allows you to make calls on your conference station to directory contacts by clicking items in a web browser. Your system administrator must set up this feature for you.

Use WebDialer with User Options Directory

Procedure

Step 1 Log in to your User Options web pages.

Step 2 Choose **User Options** > **Directory** and search for a coworker.

Step 3 Click the number that you want to dial.

Step 4 If this is your first time using WebDialer, set up preferences on the Make Call page.

Step 5 Click **Dial**.

Step 6 The call is now placed on your phone.

Step 7 To end a call, click **Hangup** or hang up from your phone.

Use WebDialer with Another Online Corporate Directory (Not User Options Directory)

Procedure

Step 1 Log in to a WebDialer-enabled corporate directory and search for coworkers.

Step 2 Click the number that you want to dial.

Step 3 When prompted, enter your user ID and password.

Step 4 If this is your first time using WebDialer, set up preferences on the Make Call page.

Step 5 Click **Dial**.
The call is now placed on your phone.

Step 6 To end a call, click **Hangup** or hang up from your phone.

Log Out of WebDialer

Procedure

Click the **Logout** icon in the Make Call or Hang Up page.

Set Up, View, or Change WebDialer Preferences

Procedure

Step 1 Access the Make Call page.
The Make Call page appears the first time that you use WebDialer (after you click the number that you want to dial).

Step 2 (Optional) Make changes to your settings.
The Make Call page contains the following options:

- Preferred language: Determines the language used for WebDialer settings and prompts.
- Use preferred device: Identifies the Cisco Unified IP Phone (Calling device) and directory number (Calling device or line) that you use to place WebDialer calls. If you have one phone with a single line, the appropriate phone and line are automatically selected. Otherwise, choose a phone or line. If you have more than one phone, it will be specified by device type and MAC address. To display the MAC address on your phone, choose **Settings** > **Network Configuration > MAC address**.

  Note If you have an Extension Mobility profile, you can select your Extension Mobility logged-in device from the Calling device menu.

- Use Extension Mobility: If selected, prompts Cisco WebDialer to use the conference station that is associated with your Cisco Extension Mobility profile (if available).
- Do not display call confirmation: If selected, prompts WebDialer to suppress the Make Call page. This page appears by default after you click a phone number in a WebDialer-enabled online directory.
- Disable Auto Close: If selected, prompts Cisco WebDialer to display the Hang Up page.

CHAPTER 8

Additional Options

Your system administrator can configure your Cisco Unified IP Conference Phone 8831 to use specific button and softkey templates along with special services and features, if appropriate. This table provides an overview of some configuration options that you might want to discuss with your system administrator based on your calling needs and work environment.

If You...	Then...	For More Information...
Need to handle more calls on the conference station line	Ask your system administrator to configure your line to support more calls. **Note** Your conference station supports a maximum of 6 calls.	Contact your system administrator or phone support team.
Need more speed dial entries	First make sure that all your speed dial entries are assigned. If you need additional speed dials, try using abbreviated dials or fast dials.	See Speed Dial, on page 61
Want to use one extension for several phones	Request a shared line. This allows you, for example, to use one extension for your desk phone and lab phone.	See Shared Lines, on page 60

If You...	Then...	For More Information...
Share conference stations or office space with coworkers	Consider using: • Call Park to store and retrieve calls without using the transfer feature. • Call Pickup to answer calls ringing on another phone. • A shared line to view or join coworkers' calls. • Cisco Extension Mobility to apply the conference station number and user profile to a shared conference station.	Contact your system administrator about these features and see the following: • Call Park, on page 43 • Call Pickup, on page 43 • Shared Lines, on page 60 • Cisco Extension Mobility, on page 50
Want to temporarily apply the conference station number and settings to a shared conference station	Ask your system administrator about the Cisco Extension Mobility Service.	See Cisco Extension Mobility, on page 50
Want to extend the area covered by audio	Ask your administrator about: • Linking to another Conference Station sound base • Adding a Wireless Extension Microphone Kit • Adding a Wired Extension Microphone Kit	See • Linked Mode, on page 23 • Enhanced Room Coverage, on page 47

CHAPTER 9

Troubleshooting

- General Troubleshooting, page 81
- Phone Administration Data, page 82
- Quality Reporting Tool, page 82

General Troubleshooting

This section provides information to help you troubleshoot general problems with the conference station. For more information, see your system administrator.

Symptom	Explanation
You cannot hear a dial tone or complete a call.	One or more of the following factors might apply: • You must log into the Cisco Extension Mobility Service. • The conference station has time-of-day restrictions that prevent you from using some features during certain hours of the day.
A conference station button is unresponsive.	Your system administrator may have disabled the button on the conference station.
The softkey that you want to use does not appear.	One or more of the following factors might apply: • You must change the line state (for example, go off-hook or have a connected call). • You must press More to reveal additional softkeys. • The conference station is not configured to support the feature associated with that softkey. Contact your administrator for more information.

Symptom	Explanation
Barge fails and results in a fast busy tone.	One or more of the following factors might apply: • You cannot barge an encrypted call if the conference station you are using is not configured for encryption. When your barge attempt fails for this reason, the conference station plays a fast busy tone. • You have toggled on the Privacy softkey.
You are disconnected from a call that you joined using Barge.	You will be disconnected from a call that you have joined using Barge if the call is put on hold, transferred, or turned into a conference call.
Cisco CallBack fails.	The other party might have call forwarding enabled.

Phone Administration Data

Your system administrator might ask you to access administration data on the conference station for troubleshooting purposes.

If You Are Asked To...	Then...
Access network configuration data	Choose **Apps** > **Admin Settings > Network Configuration**, and then navigate to the item that you want to view.
Access status data	Choose **Apps** > **Admin Settings > Status**, and then navigate to the item that you want to view.
Access model information	Choose **Apps** > **Admin Settings > Phone Information**, and then navigate to the item that you want to view.

Quality Reporting Tool

Your system administrator may temporarily configure the conference station with the Quality Reporting Tool (QRT) to troubleshoot performance problems. You can press QRT to submit information to your system administrator. Depending on your configuration, use the QRT to:

- Immediately report an audio problem on a current call.
- Select a general problem from a list of categories and choose reason codes.

CHAPTER 10

Cisco One-Year Limited Hardware Warranty Terms

- Warranty Overview, page 83
- View Cisco Information Packet, page 83
- Cisco One-Year Limited Hardware Warranty Terms, page 84

Warranty Overview

There are special terms applicable to your hardware warranty and various services that you can use during the warranty period. Your formal Warranty Statement, including the warranties and license agreements applicable to Cisco software, is available on Cisco.com.

View Cisco Information Packet

Follow these steps to access and download the Cisco Information Packet and your warranty and license agreements from Cisco.com.

Procedure

Step 1 Launch your browser, and go to this URL:
http://www.cisco.com/en/US/products/prod_warranties_listing.html

The Warranties and License Agreements page appears.

Step 2 To read the *Cisco Information Packet*, follow these steps:

a) Click the **Information Packet Number** field, and make sure that the part number 78-5235-03B0 is highlighted.
b) Select the language in which you would like to read the document.
c) Click **Go**.
d) The Cisco Limited Warranty and Software License page from the Information Packet appears.

e) Read the document online, or click the **PDF** icon to download and print the document in Adobe Portable Document Format (PDF).

Note You must have Adobe Acrobat Reader to view and print PDF files. You can download the reader from http://www.adobe.com

Step 3 To read translated and localized warranty information about your product, follow these steps:

a) Enter this part number in the Warranty Document Number field: 78-10747-01C0
b) Select the language in which you would like to view the document.
c) Click **Go**. The Cisco warranty page appears.
d) Read the document online, or click the **PDF** icon to download and print the document in Adobe Portable Document Format (PDF).

You can also contact the Cisco service and support website for assistance: http://www.cisco.com/public/Support_root.shtml

Cisco One-Year Limited Hardware Warranty Terms

Special terms apply to your hardware warranty and services that you can use during the warranty period.

Your formal Warranty Statement, including the warranties and license agreements applicable to Cisco software, is available on Cisco.com at the following URL:

http://www.cisco.com/en/US/docs/general/warranty/English/1Y1DEN__.html

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