

Cisco IP Phone 8800 Series Administration Guide for Cisco Unified Communications Manager

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Preface

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- Related Documentation, on page xiv
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Overview

The *Cisco IP Phone 8800 Series Administration Guide for Cisco Unified Communications Manager* provides the information you need to understand, install, configure, manage, and troubleshoot the phones on a VoIP network.

Because of the complexity of an IP telephony network, this guide does not provide complete and detailed information for procedures that you need to perform in Cisco Unified Communications Manager or other network devices.

Audience

Network engineers, system administrators, and telecom engineers should review this guide to learn the steps that are required to set up Cisco IP Phones. The tasks described in this document involve configuring network settings that are not intended for phone users. The tasks in this manual require a familiarity with Cisco Unified Communications Manager.

Guide Conventions

This document uses the following conventions:

Convention	Description
boldface font	Commands and keywords are in **boldface**.
italic font	Arguments for which you supply values are in *italics*.
[]	Elements in square brackets are optional.

Convention	Description
{x | y | z}	Alternative keywords are grouped in braces and separated by vertical bars.
[x | y | z]	Optional alternative keywords are grouped in brackets and separated by vertical bars.
string	A nonquoted set of characters. Do not use quotation marks around the string or the string w include the quotation marks.
`screen` font	Terminal sessions and information the system displays are in `screen` font.
`input` font	Information you must enter is in **`input`** font.
italic screen font	Arguments for which you supply values are in *italic screen* font.
^	The symbol ^ represents the key labeled Control - for example, the key combination ^D in a s display means hold down the Control key while you press the D key.
<>	Nonprinting characters such as passwords are in angle brackets.

Note Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the publication.

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Warnings use the following convention:

Attention IMPORTANT SAFETY INSTRUCTIONS

This warning symbol means danger. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents. Use the statement number provided at the end of each warning to locate its translation in the translated safety warnings that accompanied this device. Statement 1071

SAVE THESE INSTRUCTIONS

Related Documentation

Use the following sections to obtain related information.

Cisco IP Phone 8800 Series Documentation

Find documentation specific to your language, phone model, and call control system on the product support page for the Cisco IP Phone 8800 Series.

For help information about Cisco Video Phone 8875, see Cisco Video Phone 8875.

The Deployment Guide is located at the following URL:

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

Cisco Unified Communications Manager Documentation

See the *Cisco Unified Communications Manager Documentation Guide* and other publications that are specific to your Cisco Unified Communications Manager release on the product support page.

Cisco Business Edition 6000 Documentation

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

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

Documentation, Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, reviewing security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

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

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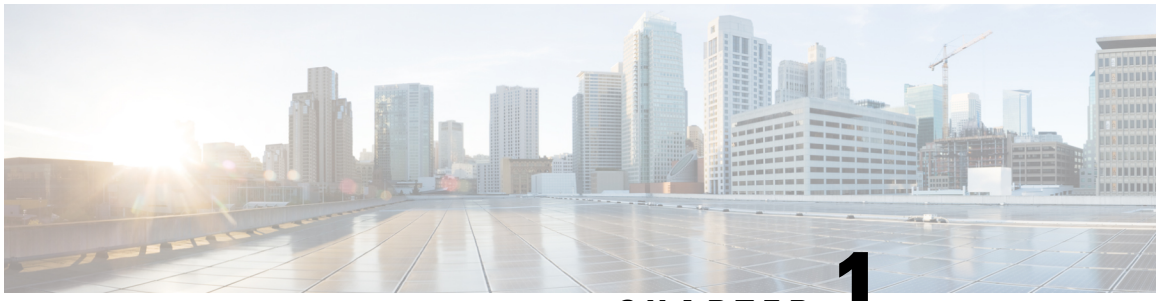

CHAPTER 1

New and Changed Information

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- New and Changed Information for Firmware Release 14.1(1), on page 2
- New and Changed Information for Firmware Release 14.0(1), on page 2
- New and Changed Information for Firmware Release 12.8(1), on page 2
- New and Changed Information for Firmware Release 12.7(1), on page 3
- New and Changed Information for Firmware Release 12.6(1), on page 3
- New Information for Firmware Release 12.5(1)SR3, on page 4
- New Information for Firmware Release 12.5(1)SR1, on page 4
- New Information for Firmware Release 12.1(1)SR1, on page 5
- New Information for Firmware Release 12.1(1), on page 5
- New Information for Firmware Release 12.0(1), on page 6
- New Information for Firmware Release 11.7(1), on page 6
- New Information for Firmware Release 11.5(1)SR1, on page 6
- New Information for Firmware Release 11.5(1), on page 7
- New Information for Firmware Release 11.0, on page 7

New and Changed Information for Firmware Release 14.2(1)

The following information is new or changed for Firmware Release 14.2(1).

Feature	New or Changed
Support for SIP OAuth on SRST	Security Enhancements for Your Phone Network, on page 81
Simplified Extension Mobility Login with Cisco Headset 730 USB Adapter	Telephone Features, on page 116
Bluetooth Mute Sync for Cisco Headset 700 Series	Telephone Features, on page 116
New settings for Cisco Headset 500 Series: Dock Event and Always On Mode	Telephone Features, on page 116

New and Changed Information for Firmware Release 14.1(1)

The following information is new or changed for Firmware Release 14.1(1).

Feature	New or Changed
SIP OAuth for Proxy TFTP support	Security Enhancements for Your Phone Network, on page 81
Improved Call Alert for Hunt Group	Telephone Features, on page 116
Configurable Calling Number Display for Enhanced Line Mode	Product Specific Configuration
Configurable Delayed PLAR	Telephone Features, on page 116
MRA Support for Extension Mobility Login with Cisco Headsets	Telephone Features, on page 116
Phone Migration without Transition Load	Migration of your Phone to a Multiplatform Phone Directly, on page 183

New and Changed Information for Firmware Release 14.0(1)

Table 1: New and Changed Information

Feature	New or Changed
Call Park Monitoring Enhancement	Product Specific Configuration, on page 136
SIP OAuth Enhancements	Security Enhancements for Your Phone Network, on page 81
User Interface Enhancements	Survivable Remote Site Telephony, on page 72 Telephone Features, on page 116
OAuth Enhancements for MRA	Mobile and Remote Access Through Expressway, on page 167

As of Firmware Release 14.0, the phones support DTLS 1.2. DTLS 1.2 requires Cisco Adaptive Security Appliance (ASA) Release 9.10 or later. You configure the minimum DTLS version for a VPN connection in ASA. For more information, see *ASDM Book 3: Cisco ASA Series VPN ASDM Configuration Guide* at https://www.cisco.com/c/en/us/support/security/asa-5500-series-next-generation-firewalls/products-installation-and-configuration-guides-list.html

New and Changed Information for Firmware Release 12.8(1)

The following information is new or changed for Firmware Release 12.8(1).

Feature	New or Changed Content
Phone Data Migration	Change a User's Phone Model, on page 63
Headset Update Enhancement	Device Information, on page 223
Simplify Extension Mobility Login with Cisco Headsets	Telephone Features, on page 116
Feature Control Changes	Product Specific Configuration, on page 136, new fields **Lower Your Voice Alert** and **Mark Call As Spam**
General changes	Clarify Wi-Fi and the PC Port: • Set Up Phone from Setup Menus, on page 47 • Enable the Wireless LAN on the Phone, on page 49
Add additional information about the Web Access field	Product Specific Configuration, on page 136
Remove unsupported feature	Telephone Features, on page 116

New and Changed Information for Firmware Release 12.7(1)

Table 2: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.7(1)

Revision	Updated Section
Updated for wallpaper support on key expansion modules.	Custom Background Images, on page 109
Updated for Cisco Headset 730 support	Device Information, on page 223
Updated for Cisco Headset 500 Series Firmware Release 2.0	Device Information, on page 223 Headset Management on Older Versions of Cisco Unified Communications Manager, on page 195
Updated for incoming hunt group calls.	Telephone Features, on page 116
E-hook configuration information was removed.	Product Specific Configuration, on page 136

New and Changed Information for Firmware Release 12.6(1)

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 3: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.6(1)

Revision	Updated Section
Updated for Revert to Primary Line in Session line mode.	Product Specific Configuration, on page 136 Session Line Mode Environment, on page 155

New Information for Firmware Release 12.5(1)SR3

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 4: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.5(1)SR3

Revision	Updated Section
Support for Activation Code Onboarding and Mobile and Remote Access	Activation Code Onboarding and Mobile and Remote Access, on page 43
Support for Problem Report Tool use from Cisco Unified Communications Manager.	Create a Phone Problem Report from Cisco Unified Communications Manager, on page 253
New topic	Share a Network Connection with Your Phone and Computer, on page 46
New topic	Protect Your Video Phone Camera, on page 36

New Information for Firmware Release 12.5(1)SR1

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 5: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.5(1)SR1

Revision	Updated Section
Support for Elliptic Curve support	Supported Security Features, on page 82
Support for Call History enhancements for Enhanced line mode with rollover lines	Features Available in Enhanced Line Mode, on page 190
Support for Whisper Paging on Cisco Unified Communications Manager Express support	Cisco Unified Communications Manager Express Interaction, on page 20
Support for Chinese language support	Language Limitation, on page 266
Support for Activation Code Onboarding	Activation Code Onboarding for On-premises Phones, on page 42

Revision	Updated Section
Support for Media Paths and Interactive Connectivity Establishment	Media Paths and Interactive Connectivity Establishment, on page 169
Support for Disable TLS Ciphers	Product Specific Configuration, on page 136
Support for Disable handset so audio path can be kept on headset	Product Specific Configuration, on page 136
Support for Remote Configuration of Headset Parameters	Headset Management on Older Versions of Cisco Unified Communications Manager, on page 195

New Information for Firmware Release 12.1(1)SR1

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 6: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.1(1)SR1

Revision	Updated Section
Enbloc Dialing for Inter-Digit Timer T.302 Enhancement.	Product Specific Configuration, on page 136

New Information for Firmware Release 12.1(1)

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 7: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.1(1)

Revision	Updated Section
Mobile and Remote Access Through Expressway now supports Enhanced line mode.	Phone Features Available for Mobile and Remote Access Through Expressway, on page 169
	Mobile and Remote Access Through Expressway, on page 167
	Features Available in Enhanced Line Mode, on page 190
Enabling or disabling TLS 1.2 for web server access is now supported.	Product Specific Configuration, on page 136
The G722.2 AMR-WB audio codec is now supported.	Phone Overview, on page 27
	Call Statistics Fields, on page 218

New Information for Firmware Release 12.0(1)

All new features have been added to Telephone Features, on page 116.

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 8: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 12.0(1)

Revision	Updated Section
Updated for Call Park, Call Park Line Status, Group Pickup, and Hunt Groups support on Enhanced Line Mode	Features Available in Enhanced Line Mode, on page 190

New Information for Firmware Release 11.7(1)

No administration updates were required for firmware release 11.7(1).

New Information for Firmware Release 11.5(1)SR1

All new features have been added to Telephone Features, on page 116.

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 9: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 11.5(1)SR1

Revision	Updated Section
Updated for Cisco IP Phone 8865NR support	• Phone Power Requirements, on page 14 • Network Protocols, on page 16 • Phone Overview, on page 27 • Buttons and Hardware, on page 34
Updated for Recording and Monitoring support on Enhanced Line Mode	Features Available in Enhanced Line Mode, on page 190
Updated for WLAN Scan List support	Enable the Wireless LAN on the Phone, on page 49
	Set Up Wireless LAN from Phone, on page 50
	Configure Network Settings, on page 55
Updated for Do not disturb with MLPP support	Set Up Do Not Disturb, on page 162
Updated for Configurable Ringer support	Product Specific Configuration, on page 136

Revision	Updated Section
Enhanced Security	Security Enhancements for Your Phone Network, on page 81
General changes	Updates to Cisco IP Phone Web Page, on page 222 New presentation of phone feature configuration in Cisco Unified Communications Manager Phone Feature Configuration, on page 135

New Information for Firmware Release 11.5(1)

Table 10: Cisco IP Phone 8800 Administration Guide revisions for Firmware Release 11.5(1).

Revision	Updated Section
Enhanced line mode is supported.	Set Up Additional Line Keys, on page 189 Features Available in Enhanced Line Mode, on page 190
Do Not Disturb (DND) was updated for new display.	Set Up Do Not Disturb, on page 162
Opus codec is supported.	Phone Overview, on page 27
FIPS Mode was added.	Enable FIPS Mode, on page 88
WLAN setup was updated.	Set Up Wireless LAN from Phone, on page 50
WLAN Profile for Cisco IP Phone 8861 and 8865 is supported.	Set Up a Wi-Fi Profile using Cisco Unified Communications Manager, on page 53
	Set Up a Wi-Fi Group using Cisco Unified Communications Manager, on page 55
Set WLAN Authentication Attempts is supported.	Set the Number of WLAN Authentication Attempts, on page 52
Enable WLAN Prompt Mode is supported.	Enable WLAN Prompt Mode, on page 53
Customize Dial Tone is supported.	Customize the Dial Tone, on page 112
Display Network Info Screen is supported.	Display Network Information Screen, on page 213

New Information for Firmware Release 11.0

All new features have been added to Telephone Features, on page 116.

All references into Cisco Unified Communications Manager documentation have been updated to support all Cisco Unified Communications Manager releases.

Table 11: Cisco IP Phone 8800 Administration Guide Revisions for Firmware Release 11.0

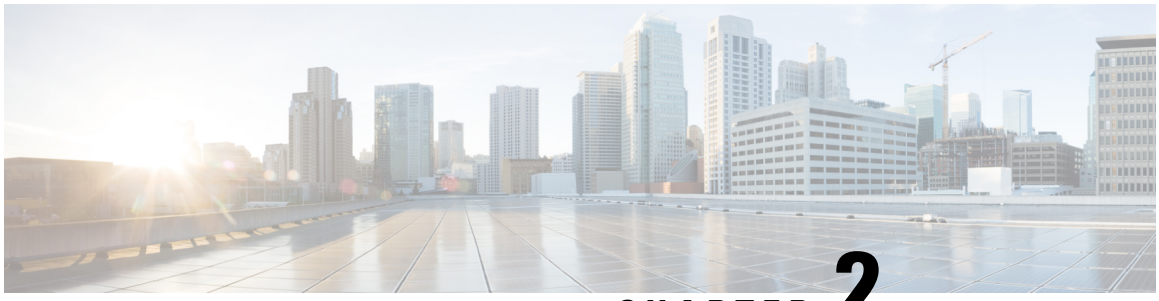
Revision	Updated Section
Updated for clarification and to address deficiencies	• VPN Configuration, on page 188 • Configure Network Settings, on page 55 • Set Up Energy Efficient Ethernet for Switch and PC Port, on page 166 • Video Transmit Resolution Setup, on page 193 • Enhanced Survivable Remote Site Telephony, on page 75
Updated for improved Sectional phone debug option support	• Control Debug Information from Cisco Unified Communications Manager, on page 257.
Updated for improved EAP-TLS + SCEP, PEAP-GTC, and X.509 digital certificates support	• WLAN Security, on page 91. • Set Up Authentication Mode, on page 94 • Wireless Security Credentials, on page 94
Updated for improved Problem Report Tool(PRT) support	• Problem Report Tool, on page 172. • Configure a Customer Support Upload URL, on page 173.
Added for Application Dial Rule support	• Application Dial Rules, on page 75
Added for Line Text Label	• Set the Label for a Line, on page 174.



PART I

About the Cisco IP Phone

- Technical Details, on page 11
- Cisco IP Phone Hardware, on page 27



CHAPTER 2

Technical Details

- Physical and Operating Environment Specifications, on page 11
- Cable Specifications, on page 12
- Phone Power Requirements, on page 14
- Network Protocols, on page 16
- VLAN Interaction, on page 19
- Cisco Unified Communications Manager Interaction, on page 20
- Cisco Unified Communications Manager Express Interaction, on page 20
- Voice Messaging System Interaction, on page 21
- Phone Startup Overview, on page 21
- External Devices, on page 23
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- Phone Configuration Files, on page 24
- Phone Behavior During Times of Network Congestion, on page 24
- Phone Behavior on a Network with Two Network Routers, on page 25
- Application Programming Interface, on page 25

Physical and Operating Environment Specifications

The following table shows the physical and operating environment specifications for the Cisco IP Phone 8800 Series.

Table 12: Physical and Operating Specifications

Specification	Value or range
Operating temperature	32° to 104°F (0° to 40°C)
Operating relative humidity	Operating: 10% to 90% (non-condensing) Non-operating: 10% to 95% (non-condensing)
Storage temperature	14° to 140°F (–10° to 60°C)
Height	9.02 in. (229.1 mm)
Width	10.13 in. (257.34 mm)

Specification	Value or range
Depth	1.57 in. (40 mm)
Weight	2.62 lb (1.19 kg)
Power	100-240 VAC, 50-60 Hz, 0.5 A when using the AC adapter 48 VDC, 0.2 A when using the in-line power over the network cable
Cables	Category 3/5/5e/6 for 10-Mbps cables with 4 pairs Category 5/5e/6 for 100-Mbps cables with 4 pairs Category 5e/6 for 1000-Mbps cables with 4 pairs **Note** Cables have 4 pairs of wires for a total of 8 conductors.
Distance requirements	As supported by the Ethernet Specification, the maximum cable length betwee Cisco IP Phone and the switch is assumed to be 330 feet (100 meters).

Cable Specifications

The following information lists the cable specifications:

- RJ-9 jack (4-conductor) for handset and headset connection
- RJ-45 jack for the LAN 10/100/1000BaseT connection (10/100/1000 Network port on the phone)
- RJ-45 jack for a second 10/100/1000BaseT compliant connection (10/100/1000 Computer port on the phone)
- 3.5 mm jack for speaker connection (only Cisco IP Phone 8861)
- 48-volt power connector
- USB ports/connector: one USB port for Cisco IP Phone 8851 and two USB ports for Cisco IP Phone 8861
- 3 key expansion modules connectors which is considered as USB connector for Cisco IP Phone 8851 and 8861

Network and Computer Port Pinouts

Although both the network and computer (access) ports are used for network connectivity, they serve different purposes and have different port pinouts.

- The network port is the 10/100/1000 SW port on the Cisco IP Phone.
- The computer (access) port is the 10/100/1000 PC port on the Cisco IP Phone.

Network Port Connector

The following table describes the network port connector pinouts.

Table 13: Network Port Connector Pinouts

Pin Number	Function
1	BI_DA+
2	BI_DA-
3	BI_DB+
4	BI_DC+
5	BI_DC-
6	BI_DB-
7	BI_DD+
8	BI_DD-
Note	BI stands for bidirectional, while DA, DB, DC, and DD stand for Data A, Data B, Data C, and Data D respectively.

Computer Port Connector

The following table describes the computer port connector pinouts.

Table 14: Computer (Access) Port Connector Pinouts

Pin Number	Function
1	BI_DB+
2	BI_DB-
3	BI_DA+
4	BI_DD+
5	BI_DD-
6	BI_DA-
7	BI_DC+
8	BI_DC-
Note	BI stands for bidirectional, while DA, DB, DC, and DD stand for Data A, Data B, Data C, and Data D respectively.

Phone Power Requirements

The Cisco IP Phone can be powered with external power or with Power over Ethernet (PoE). A separate power supply provides external power. The switch can provide PoE through the phone Ethernet cable.

Cisco IP Phones 8861 and 8865 are PoE Class 4 devices and require a switch or line card with Class 4 capabilities to support extra features.

For more information on your phone's power requirements, consult your phone's data sheet.

When you install a phone that is powered with external power, connect the power supply before you connect the Ethernet cable to the phone. When you remove a phone that is powered with external power, disconnect the Ethernet cable from the phone before you disconnect the power supply.

Table 15: Guidelines for Cisco IP Phone Power

Power type	Guidelines
External power: Provided through the CP-PWR-CUBE-4= external power supply	The Cisco IP Phone uses the CP-PWR-CUBE-4 power supply.
PoE power—Provided by a switch through the Ethernet cable attached to the phone.	Cisco IP Phones 8851, 8851NR, 8861, 8865, and 8865NR support 802.3at PoE for a use. For more information, consult your phone's data sheet. The switch requires a backup power supply for uninterruptible operation of the phon Make sure that the CatOS or IOS version that runs on your switch supports your intend deployment. See the documentation for your switch for operating system version inf
Universal Power over Ethernet (UPoE)	Cisco IP Phones 8865 and 8865NR supports UPoE.

The documents in the following table provide more information on the following topics:

- Cisco switches that work with Cisco IP Phones
- Cisco IOS releases that support bidirectional power negotiation
- Other requirements and restrictions about power

Table 16: Additional Information

Document topics	URL
PoE Solutions	http://www.cisco.com/c/en/us/solutions/enterprise-networks/power-over-ethernet-solutions/index.html
UPoE	http://www.cisco.com/c/en/us/solutions/enterprise-networks/upoe
Cisco Catalyst Switches	http://www.cisco.com/c/en/us/products/switches/index.html
Integrated Service Routers	http://www.cisco.com/c/en/us/products/routers/index.html
Cisco IOS Software	http://www.cisco.com/c/en/us/products/ios-nx-os-software/index.

Power Outage

Your access to emergency service through the phone requires that the phone receive power. If a power interruption occurs, service or emergency calling service dialing does not function until power is restored. If a power failure or disruption occurs, you may need to reset or reconfigure the equipment before you can use service or emergency calling service dialing.

Power Reduction

You can reduce the amount of energy that the Cisco IP Phone consumes by using Power Save or EnergyWise (Power Save Plus) mode.

Power Save

In Power Save mode, the backlight on the screen is not lit when the phone is not in use. The phone remains in Power Save mode for the scheduled duration or until the user lifts the handset or presses any button.

Power Save Plus (EnergyWise)

The Cisco IP Phone supports Cisco EnergyWise (Power Save Plus) mode. When your network contains an EnergyWise (EW) controller (for example, a Cisco switch with the EnergyWise feature enabled), you can configure these phones to sleep (power down) and wake (power up) on a schedule to further reduce power consumption.

Set up each phone to enable or disable the EnergyWise settings. If EnergyWise is enabled, configure a sleep and wake time, as well as other parameters. These parameters are sent to the phone as part of the phone configuration XML file.

Power Negotiation Over LLDP

The phone and the switch negotiate the power that the phone consumes. Cisco IP Phone operates at multiple power settings, which lowers power consumption when less power is available.

After a phone reboots, the switch locks to one protocol (CDP or LLDP) for power negotiation. The switch locks to the first protocol (containing a power Threshold Limit Value [TLV]) that the phone transmits. If the system administrator disables that protocol on the phone, the phone cannot power up any accessories because the switch does not respond to power requests in the other protocol.

Cisco recommends that Power Negotiation always be enabled (default) when connecting to a switch that supports power negotiation.

If Power Negotiation is disabled, the switch may disconnect power to the phone. If the switch does not support power negotiation, disable the Power Negotiation feature before you power up accessories over PoE. When the Power Negotiation feature is disabled, the phone can power the accessories up to the maximum that the IEEE 802.3af-2003 standard allows.



Note

- When CDP and Power Negotiation are disabled, the phone can power the accessories up to 15.4W.

Network Protocols

Cisco IP Phone 8800 Series support several industry-standard and Cisco network protocols required for voice communication. The following table provides an overview of the network protocols that the phones support.

Table 17: Supported Network Protocols on the Cisco IP Phone 8800 Series

Network protocol	Purpose	Usage notes
Bluetooth	Bluetooth is a wireless personal area network (WPAN) protocol that specifies how devices communicate over short distances.	Cisco IP Phones 8845, 8865, and 8851 support Bluetooth 4.1. Cisco IP Phone 8861 support Bluetooth 4.0. Cisco IP Phone 8811, 8841, 8851NR, and 8865NR do not support Bluetooth.
Bootstrap Protocol (BootP)	BootP enables a network device, such as the Cisco IP Phone, to discover certain startup information, such as the IP address.	—
Cisco Audio Session Tunnel (CAST)	The CAST protocol allows your phones and associated applications to communicate with the remote IP Phones without requiring changes to the signaling components.	The Cisco IP Phone uses CAST as an interface between CUVA and Cisco Unified Communications Manager using the Cisco IP Phone as a SIP proxy.
Cisco Discovery Protocol (CDP)	CDP is a device-discovery protocol that runs on all Cisco-manufactured equipment. Using CDP, a device can advertise its existence to other devices and receive information about other devices in the network.	The Cisco IP Phones use CDP to communicate information such as auxiliary VLAN ID, per port power management details, and Quality of Service (QoS) configuration information with the Cisco Catalyst switch.
Cisco Peer-to-Peer Distribution Protocol (CPPDP)	CPPDP is a Cisco proprietary protocol used to form a peer-to-peer hierarchy of devices. This hierarchy is used to distribute firmware files from peer devices to their neighboring devices.	CPPDP is used by the Peer Firmware Sharing feature.
Dynamic Host Configuration Protocol (DHCP)	DHCP dynamically allocates and assigns an IP address to network devices. DHCP enables you to connect an IP phone into the network and the phone to become operational without the need to manually assign an IP address or to configure additional network parameters.	DHCP is enabled by default. If disabled, you must manually configure the IP address, subnet mask, gateway, and a TFTP server on each phone locally. We recommend that you use DHCP custom option 150. With this method, you configure the TFTP server IP address as the option value. For more information, see the documentation for your particular Cisco Unified Communications Manager release. **Note** If you cannot use option 150, you may try using DHCP option 66.

Network protocol	Purpose	Usage notes
Hypertext Transfer Protocol (HTTP)	HTTP is the standard way of transferring information and moving documents across the Internet and the web.	Cisco IP Phones use HTTP for XML services and for troubleshooting purposes.
Hypertext Transfer Protocol Secure (HTTPS)	Hypertext Transfer Protocol Secure (HTTPS) is a combination of the Hypertext Transfer Protocol with the SSL/TLS protocol to provide encryption and secure identification of servers.	Web applications with both HTTP and HTTPS support have two URLs configured. Cisco IP Phones that support HTTPS choose the HTTPS URL.
IEEE 802.1X	The IEEE 802.1X standard defines a client-server-based access control and authentication protocol that restricts unauthorized clients from connecting to a LAN through publicly accessible ports. Until the client is authenticated, 802.1X access control allows only Extensible Authentication Protocol over LAN (EAPOL) traffic through the port to which the client is connected. After authentication is successful, normal traffic can pass through the port.	The Cisco IP Phone implements the IEEE 802.1X standard by providing support for the following authentication methods: EAP-FAST, and EAP-TLS. When 802.1X authentication is enabled on the phone, you should disable the PC port and voice VLAN.
IEEE 802.11n/802.11ac	The IEEE 802.11 standard specifies how devices communication over a wireless local area network (WLAN). 802.11n operates at the 2.4 GHz and 5 GHz band and 802.11ac operates at the 5 GHz band.	The 802.11 interface is a deployment option for cases when Ethernet cabling is unavailable or undesirable. Only Cisco IP Phone 8861 and 8865 support WLAN.
Internet Protocol (IP)	IP is a messaging protocol that addresses and sends packets across the network.	To communicate using IP, network devices must have an assigned IP address, subnet, and gateway. IP addresses, subnets, and gateway identifications are automatically assigned if you are using the Cisco IPPhone with Dynamic Host Configuration Protocol (DHCP). If you are not using DHCP, you must manually assign these properties to each phone locally. The Cisco IP Phones support IPv6 addresses. For more information, see the documentation for your particular Cisco Unified Communications Manager release.
Link Layer Discovery Protocol (LLDP)	LLDP is a standardized network discovery protocol (similar to CDP) that is supported on some Cisco and third-party devices.	The Cisco IPPhone supports LLDP on the PC port.

Network protocol	Purpose	Usage notes
Link Layer Discovery Protocol-Media Endpoint Devices (LLDP-MED)	LLDP-MED is an extension of the LLDP standard for voice products.	The Cisco IPPhone supports LLDP-MED on the SW port to communicate information such as: • Voice VLAN configuration • Device discovery • Power management • Inventory management
Real-Time Transport Protocol (RTP)	RTP is a standard protocol for transporting real-time data, such as interactive voice, over data networks.	Cisco IP Phones use the RTP protocol to send and receive real-time voice traffic from other phones and gateways.
Real-Time Control Protocol (RTCP)	RTCP works in conjunction with RTP to provide QoS data (such as jitter, latency, and round-trip delay) on RTP streams.	RTCP is enabled by default.
Session Description Protocol (SDP)	SDP is the portion of the SIP protocol that determines which parameters are available during a connection between two endpoints. Conferences are established by using only the SDP capabilities that all endpoints in the conference support.	SDP capabilities, such as codec types, DTMF detection, and comfort noise, are normally configured on a global basis by Cisco Unified Communications Manager or Media Gateway in operation. Some SIP endpoints may allow configuration of these parameters on the endpoint itself.
Session Initiation Protocol (SIP)	SIP is the Internet Engineering Task Force (IETF) standard for multimedia conferencing over IP. SIP is an ASCII-based application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints.	Like other VoIP protocols, SIP addresses the functions of signaling and session management within a packet telephony network. Signaling allows transportation of call information across network boundaries. Session management provides the ability to control the attributes of an end-to-end call. Cisco IP Phones support the SIP protocol when the phones are operating in IPv6-only, IPv4-only, or in both IPv4 and IPv6.
Transmission Control Protocol (TCP)	TCP is a connection-oriented transport protocol.	Cisco IP Phones use TCP to connect to Cisco Unified Communications Manager and to access XML services.
Transport Layer Security (TLS)	TLS is a standard protocol for securing and authenticating communications.	Upon security implementation, Cisco IP Phones use the TLS protocol when securely registering with Cisco Unified Communications Manager.

Network protocol	Purpose	Usage notes
Trivial File Transfer Protocol (TFTP)	TFTP allows you to transfer files over the network. On the Cisco IPPhone, TFTP enables you to obtain a configuration file specific to the phone type.	TFTP requires a TFTP server in your network that the DHCP server can automatically identify. If you want a phone to use a TFTP server other than the one that the DHCP server specifies, you must manually assign the IP address of the TFTP server by using the Network Configuration menu on the phone. For more information, see the documentation for your particular Cisco Unified Communications Manager release.
User Datagram Protocol (UDP)	UDP is a connectionless messaging protocol for delivery of data packets.	UDP is used only for RTP streams. SIP signaling on the phones do not support UDP.

For more information about LLDP-MED support, see the LLDP-MED and Cisco Discovery Protocol white paper:

http://www.cisco.com/en/US/tech/tk652/tk701/technologies_white_paper0900aecd804cd46d.shtml

Related Topics

802.1X Authentication, on page 106
Configure Network Settings
Phone Startup Verification, on page 62
VLAN Interaction, on page 19
Cisco Unified Communications Manager Interaction, on page 20
Cisco Unified Communications Manager Express Interaction, on page 20
Set Up the Audio and Video Port Range, on page 177
Cisco Unified Communications Manager Documentation, on page xv

VLAN Interaction

The Cisco IP Phone contains an internal Ethernet switch, enabling forwarding of packets to the phone, and to the computer (access) port and the network port on the back of the phone.

If a computer is connected to the computer (access) port, the computer and the phone share the same physical link to the switch and share the same port on the switch. This shared physical link has the following implications for the VLAN configuration on the network:

- The current VLANs might be configured on an IP subnet basis. However, additional IP addresses might not be available to assign the phone to the same subnet as other devices that connect to the same port.
- Data traffic present on the VLAN supporting phones might reduce the quality of VoIP traffic.
- Network security may indicate a need to isolate the VLAN voice traffic from the VLAN data traffic.

You can resolve these issues by isolating the voice traffic onto a separate VLAN. The switch port to which the phone connects would be configured for separate VLANs for carrying:

- Voice traffic to and from the IP phone (auxiliary VLAN on the Cisco Catalyst 6000 series, for example)

- Data traffic to and from the PC that connects to the switch through the computer (access) port of the IP phone (native VLAN)

Isolating the phones on a separate, auxiliary VLAN increases the quality of the voice traffic and allows a large number of phones to be added to an existing network that does not have enough IP addresses for each phone.

For more information, see the documentation that is included with a Cisco switch. You can also access switch information at this URL:

http://cisco.com/en/US/products/hw/switches/index.html

Cisco Unified Communications Manager Interaction

Cisco Unified Communications Manager is an open, industry-standard call processing system. Cisco Unified Communications Manager software sets up and tears down calls between phones, integrating traditional PBX functionality with the corporate IP network. Cisco Unified Communications Manager manages the components of the telephony system, such as the phones, the access gateways, and the resources necessary for features such as call conferencing and route planning. Cisco Unified Communications Manager also provides:

- Firmware for phones
- Certificate Trust List (CTL) and Identity Trust List (ITL) files using the TFTP and HTTP services
- Phone registration
- Call preservation, so that a media session continues if signaling is lost between the primary Communications Manager and a phone

For information about configuring Cisco Unified Communications Manager to work with the phones described in this chapter, see the documentation for your particular Cisco Unified Communications Manager release.



Note If the phone model that you want to configure does not appear in the Phone Type drop-down list in Cisco Unified Communications Manager Administration, install the latest device package for your version of Cisco Unified Communications Manager from Cisco.com.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Cisco Unified Communications Manager Express Interaction

When your phone works with the Cisco Unified Communications Manager Express (Unified CME), it must go into CME mode.

When a user invokes the conference feature, the tag allows the phone to use either a local or network hardware conference bridge.

The phones do not support the following actions:

- Transfer—Only supported in the connected call transfer scenario.
- Conference—Only supported in the connected call transfer scenario.

- Join—Supported using the Conference button or hookflash access.
- Hold—Supported using the Hold button.
- Barge and Merge—Not supported.
- Direct Transfer—Not supported.
- Select—Not supported.

The users cannot create conference and transfer calls across different lines.

Unified CME supports intercom calls, also known as whisper paging. But the page is rejected by the phone during calls.

Both Session line mode and Enhanced line mode are supported in CME mode.

Voice Messaging System Interaction

Cisco Unified Communications Manager lets you integrate with different voice messaging systems, including the Cisco Unity Connection voice messaging system. Because you can integrate with various systems, you must provide users with information about how to use your specific system.

To enable the ability for a user to transfer to voicemail, set up a *xxxxx dialing pattern and configure it as Call Forward All to Voicemail. For more information, see the Cisco Unified Communications Manager documentation.

Provide the following information to each user:

- How to access the voice messaging system account.

  Make sure that you have used the Cisco Unified Communications Manager to configure the Messages button on the Cisco IP Phone.

- Initial password for accessing the voice messaging system.

  Configure a default voice messaging system password for all users.

- How the phone indicates that voice messages are waiting.

  Use Cisco Unified Communications Manager to set up a message waiting indicator (MWI) method.

Phone Startup Overview

When connecting to the VoIP network, the Cisco IP Phones goes through a standard startup process. Depending on your specific network configuration, only some of these steps may occur on your Cisco IP Phone.

1. Obtain power from the switch. If a phone is not using external power, the switch provides inline power through the Ethernet cable that is attached to the phone.
2. (For the Cisco IP Phone 8861 and 8865 in a wireless LAN only) Scan for an access point. The Cisco IP Phone 8861 and 8865 scans the RF coverage area with the radio. The phone searches the network profiles and scans for access points that contain a matching SSID and authentication type. The phone associates with the access point with the highest RSSI that matches with the network profile.

3. (For the Cisco IP Phone 8861 and 8865 in a wireless LAN only) Authenticate with the access point. The Cisco IP Phone begins the authentication process. The following table describes the authentication process:

Authentication type	Key management options	Description
Open	None	Any device can authenticate to the access point. For added security, static WEP encryption might optionally be used.
Shared Key	None	The phone encrypts the challenge text by using the WEP key and the access point must verify the WEP key that was used to encrypt the challenge text before network access is available.
PEAP or EAP-FAST	None	The RADIUS server authenticates the username and password before network access is available.

4. Load the stored phone image. At startup, the phone runs a bootstrap loader that loads a phone firmware image that is stored in flash memory. Using this image, the phone initializes the software and hardware.

5. Configure the VLAN. If the Cisco IP Phone is connected to a Cisco Catalyst switch, the switch next informs the phone of the voice VLAN that is defined on the switch. The phone needs to know the VLAN membership before it can proceed with the Dynamic Host Configuration Protocol (DHCP) request for an IP address.

6. Obtain an IP address. If the Cisco IP Phone is using DHCP to obtain an IP address, the phone queries the DHCP server to obtain one. If you are not using DHCP in your network, you must assign static IP addresses to each phone locally.

7. Request the CTL file. The TFTP server stores the CTL file. This file contains the certificates that are necessary for establishing a secure connection between the phone and Cisco Unified Communications Manager.

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

8. Request the ITL file. The phone requests the ITL file after it requests the CTL file. The ITL file contains the certificates of the entities that the phone can trust. The certificates are used to authenticate a secure connection with the servers or to authenticate a digital signature signed by the servers. Cisco Unified Communications Manager 8.5 and later supports the ITL file.

9. Access a TFTP server. In addition to assigning an IP address, the DHCP server directs the Cisco IP Phone to a TFTP Server. If the phone has a statically defined IP address, you must configure the TFTP server locally on the phone; the phone then contacts the TFTP server directly.



Note You can also assign an alternate TFTP server to use instead of the one that DHCP assigns.

10. Request the configuration file. The TFTP server has configuration files, which define parameters for connecting to Cisco Unified Communications Manager and other information for the phone.

11. Contact Cisco Unified Communications Manager. The configuration file defines how the Cisco IP Phone communicates with Cisco Unified Communications Manager and provides a phone with the load

ID. After it obtains the file from the TFTP server, the phone attempts to make a connection to the highest priority Cisco Unified Communications Manager on the list.

If the security profile of the phone is configured for secure signaling (encrypted or authenticated) and the Cisco Unified Communications Manager is set to secure mode, the phone makes a TLS connection. Otherwise, the phone makes a nonsecure TCP connection.

If the phone was manually added to the database, Cisco Unified Communications Manager identifies the phone. If the phone was not manually added to the database and autoregistration is enabled in Cisco Unified Communications Manager, the phone attempts to autoregister itself in the Cisco Unified Communications Manager database.

Note Autoregistration is disabled when you configure the CTL client. In this case, you must add the phone to the Cisco Unified Communications Manager database manually.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

External Devices

We recommend that you use good-quality external devices that are shielded against unwanted radio frequency (RF) and audio frequency (AF) signals. External devices include headsets, cables, and connectors.

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, we recommend 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].

USB Port Information

The Cisco IP Phones 8851, 8851NR, 8861, 8865, and 8865NR support a maximum of five devices that connect to each USB port. Each device that connects to the phone is included in the maximum device count. For example, your phone can support five USB devices on the side port and five more standard USB devices on

the back port. Many third-party USB products count as multiple USB devices; for example, a device containing a USB hub and headset can count as two USB devices. For more information, see the USB device documentation.

Note

- Unpowered hubs are not supported, and powered hubs with more than four ports are not supported.
- USB headsets that connect to the phone through a USB hub are not supported.

Each key expansion module connects to the phone counts as a USB device. If three key expansion modules are connected to the phone, these count as three USB devices.

Phone Configuration Files

Configuration files for a phone are stored on the TFTP server and define parameters for connecting to Cisco Unified Communications Manager. In general, any time you make a change in Cisco Unified Communications Manager that requires the phone to be reset, a change is automatically made to the phone configuration file.

Configuration files also contain information about which image load the phone should be running. If this image load differs from the one currently loaded on a phone, the phone contacts the TFTP server to request the required load files.

If you configure security-related settings in Cisco Unified Communications Manager Administration, the phone configuration file will contain sensitive information. To ensure the privacy of a configuration file, you must configure it for encryption. For more information, see the documentation for your particular Cisco Unified Communications Manager release. A phone requests a configuration file whenever it resets and registers with Cisco Unified Communications Manager.

A phone accesses a default configuration file named XmlDefault.cnf.xml from the TFTP server when the following conditions exist:

- You have enabled autoregistration in Cisco Unified Communications Manager
- The phone has not been added to the Cisco Unified Communications Manager database
- The phone is registering for the first time

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Phone Behavior During Times of Network Congestion

Anything that degrades network performance can affect phone audio 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.

Phone Behavior on a Network with Two Network Routers

The Cisco IP Phone 8800 Series uses a firewall to provide protection against cyber intrusions, such as a man-in-the-middle attack. This firewall cannot be disabled. But it could stop traffic on a phone, if you configure your network with two network routers in the same subnet and with IP redirect.

The phone firewall stops traffic because this network setup is similar to a man-in-the-middle attack. The phone receives redirect packets for different destination IPs in a different subnet from the phone. The phone is on a network with more than one router, and the default router sends traffic to a second router.

Look at the phone logs if you suspect the firewall is stopping traffic. Look for an error code 1 notification from the operating system when it tried to establish a connection. One of the signatures is

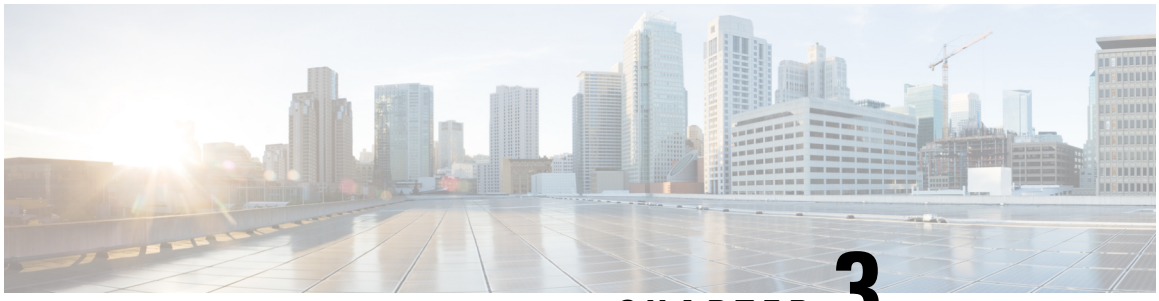
```
sip_tcp_create_connection: socket connect failed cpr_errno: 1.
```

A network with two network routers in the same subnet and with IP redirect is not a common configuration. If you are using this network setup, consider using only one router on a subnet. But if you require two network routers on the same subnet, disable IP Redirect on the router and reboot the phone.

Application Programming Interface

Cisco supports phone API utilization by 3rd party applications that have been tested and certified through Cisco by the 3rd party application developer. Any phone issues related to uncertified application interaction must be addressed by the 3rd party and will not be addressed by Cisco.

For support model of Cisco certified 3rd party applications/solutions, please refer to Cisco Solution Partner Program website for details.



CHAPTER 3

Cisco IP Phone Hardware

- Phone Overview, on page 27
- Cisco IP Phone 8811, on page 29
- Cisco IP Phones 8841 and 8845, on page 30
- Cisco IP Phones 8851 and 8851NR, on page 31
- Cisco IP Phones 8861, 8865, and 8865NR, on page 33
- Buttons and Hardware, on page 34
- Protect Your Video Phone Camera, on page 36

Phone Overview

The Cisco IP Phones 8800 Series provides voice communication over an Internet Protocol (IP) network. The Cisco IP Phone functions much like any digital business phone, allowing you to make phone calls and to access features such as mute, hold, transfer, and more. In addition, because the phone connects to your data network, it offers enhanced IP telephony features, including access to network information and services, and customizable features and services.

The Cisco IP Phone 8811 has a grayscale LCD screen. The Cisco IP Phones 8841, 8845, 8851, 8851NR, 8861, 8865, and 8865NR have a 24-bit color LCD screen.

When adding features to the phone line keys, you are limited by the number of line keys available. You cannot add more features than the number of line keys on your phone.

The Cisco IP Phones have the following features:

- Programmable feature buttons that support up to 5 lines in Session Line Mode or up to 10 lines with Enhanced Line Mode
- Full video capabilities (Cisco IP Phones 8845, 8865, and 8865NR only)
- Gigabit Ethernet connectivity
- Bluetooth support for wireless headsets (Cisco IP Phone 8845, 8851, 8861, and 8865 only. This feature is not supported on Cisco IP Phone 8811, 8841, 8851NR, and 8865NR.)
- Support for an external microphone and speakers (Cisco IP Phone 8861, 8865, and 8865NR only)
- Network connectivity by Wi-Fi (Cisco IP Phone 8861 and 8865 only. Wi-Fi is not supported on Cisco IP Phone 8865NR.)
- USB ports:

- One USB port for Cisco IP Phone 8851 and 8851NR
- Two USB ports for Cisco IP Phone 8861, 8865, and 8865NR

The Cisco IP Phone 8845, 8865, and 8865NR support video calls with a built-in video camera. Use this feature to collaborate with friends and co-workers or to hold face-to-face meetings over the phone.



Note You should save the box and packaging for the Cisco IP Phone 8845, 8865, and 8865NR. The cameras on these phones are fragile. If you move the phone, we recommend that you pack the phone into the original box to protect the camera. For more information, see Protect Your Video Phone Camera, on page 36.

A video call includes the following features:

- PIP — Select from four positions: Right bottom, Right top, Left top, and Left bottom. You can also turn PIP off.
- Swap — Toggles the views in the PIP view. The Swap softkey is disabled when PIP is off.
- Self-view Video — Select Self-view Video to view your image as it appears on video.
- Video UI and Conference/Transfer Initiation — Select to begin a conference.

For additional information on Video Calls, see *Cisco IP Phone 8800 Series User Guide for Cisco Unified Communications Manager* and the documentation for your particular Cisco Unified Communications Manager release.

Like other devices, a Cisco IP Phone must be configured and managed. These phones encode and decode the following codecs:

- G.711 a-law
- G.711 mu-law
- G.722
- G722.2 AMR-WB
- G.729a/G.729ab
- G.726
- iLBC
- Opus
- iSAC



Caution Using a cell, mobile, or GSM phone, or two-way radio in close proximity to a Cisco IP Phone might cause interference. For more information, see the manufacturer's documentation of the interfering device.

Cisco IP Phones provide traditional telephony functionality, such as call forwarding and transferring, redialing, speed dialing, conference calling, and voice messaging system access. Cisco IP Phones also provide a variety of other features.

As with other network devices, you must configure Cisco IP Phones to prepare them to access Cisco Unified Communications Manager and the rest of the IP network. By using DHCP, you have fewer settings to configure on a phone. If your network requires it, however, you can manually configure information such as: an IP address, TFTP server, and subnet information.

Cisco IP Phones can interact with other services and devices on your IP network to provide enhanced functionality. For example, you can integrate Cisco Unified Communications Manager with the corporate Lightweight Directory Access Protocol 3 (LDAP3) standard directory to enable users to search for coworker contact information directly from their IP phones. You can also use XML to enable users to access information such as weather, stocks, quote of the day, and other web-based information.

Finally, because the Cisco IP Phone is a network device, you can obtain detailed status information from it directly. This information can assist you with troubleshooting any problems users might encounter when using their IP phones. You can also obtain statistics about an active call or firmware versions on the phone.

To function in the IP telephony network, the Cisco IP Phone must connect to a network device, such as a Cisco Catalyst switch. You must also register the Cisco IP Phone with a Cisco Unified Communications Manager system before sending and receiving calls.

Related Topics

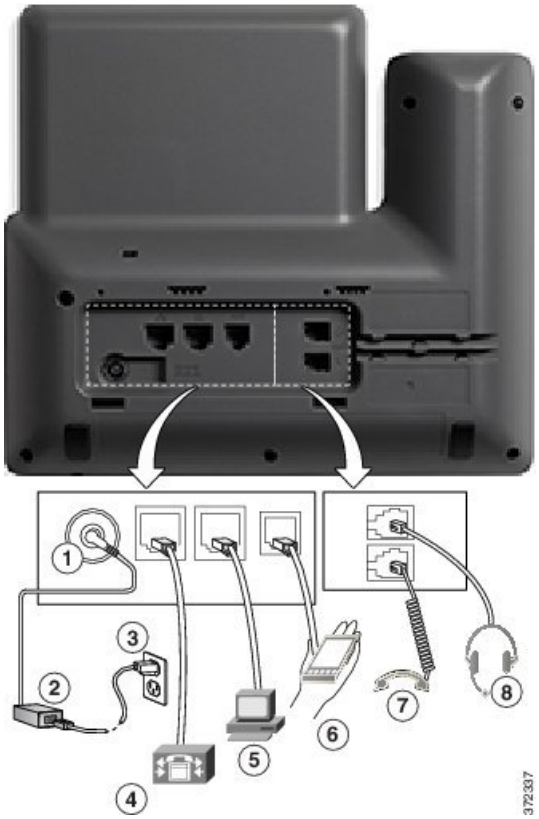
Cisco Unified Communications Manager Documentation, on page xv

Cisco IP Phone 8811

The following section describe the Cisco IP Phone 8811 attributes.

Phone Connections

Connect your phone to your organization's IP telephony network as shown in the following diagram.



1	DC adapter port (DC48V).	5	Access port (10/100/1000 PC) connection.
2	AC-to-DC power supply (optional).	6	Auxiliary port.
3	AC power wall plug (optional).	7	Handset connection.
4	Network port (10/100/1000 SW) connection. IEEE 802.3at power enabled.	8	Analog headset connection (optional).

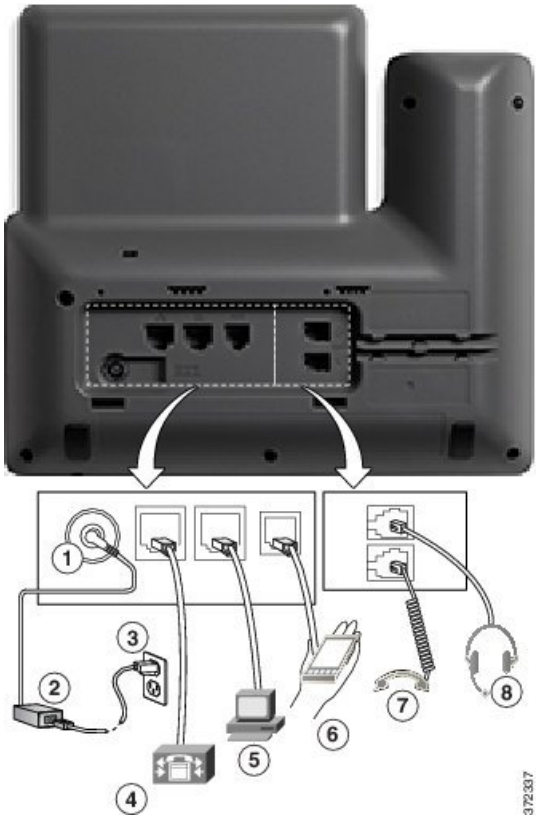
Note The Cisco IP Phone 8811 does not support a key expansion module.

Cisco IP Phones 8841 and 8845

The following section describe the attributes of the Cisco IP Phones 8841 and 8845.

Phone Connections

Connect your phone to the corporate IP telephony network, using the following diagram.



1	DC adaptor port (DC48V).	5	Access port (10/100/1000 PC) connection.
2	AC-to-DC power supply (optional).	6	Auxiliary port.
3	AC power wall plug (optional).	7	Handset connection.
4	Network port (10/100/1000 SW) connection. IEEE 802.3at power enabled.	8	Analog headset connection (optional).

Note The Cisco IP Phone 8841 and 8845 does not support a key expansion module.

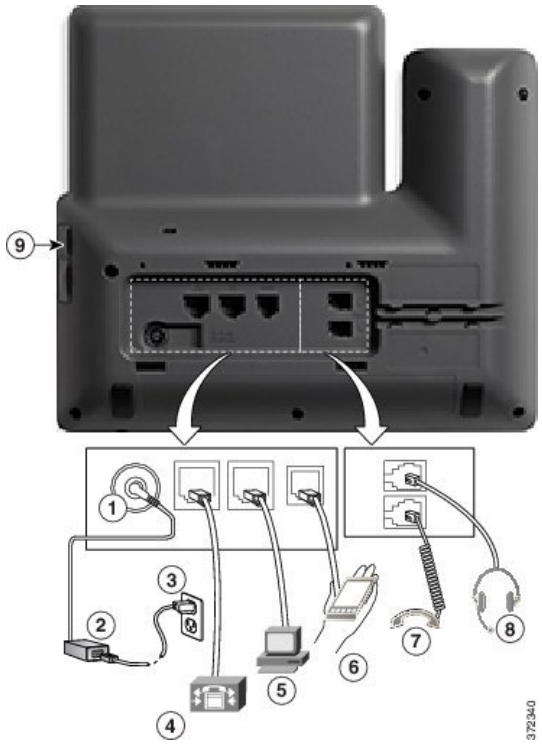
Cisco IP Phones 8851 and 8851NR

The following section describe the attributes of the Cisco IP Phones 8851 and 8851NR.

Note The Cisco IP Phone 8851NR does not support Bluetooth. Otherwise, the Cisco IP Phone 8851 and Cisco IP Phone 8851NR support the same features.

Phone Connections

Connect your phone to the corporate IP telephony network as shown in the following diagram.



1	DC adaptor port (DC48V).	6	Auxiliary port.
2	AC-to-DC power supply (optional).	7	Handset connection.
3	AC power wall plug (optional).	8	Analog headset connection (optional).
4	Network port (10/100/1000 SW) connection. IEEE 802.3at power enabled.	9	USB port
5	Access port (10/100/1000 PC) connection.		

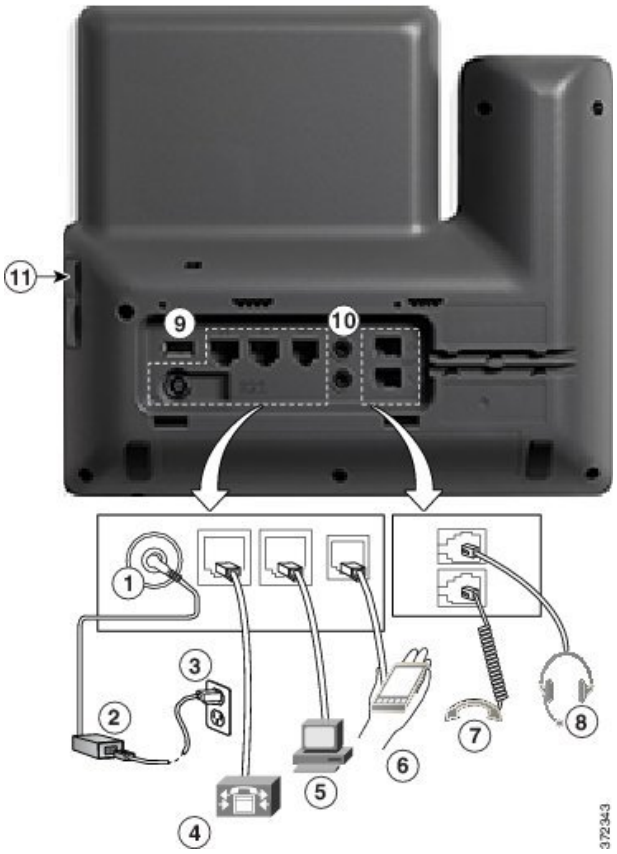
Note Each USB port supports the connection of up to five supported and nonsupported devices. Each device connected to the phone is included in the maximum device count. For example, your phone can support five USB devices (such as two key expansion modules, one headset, one hub, and one other standard USB device) on the side port. Many third-party USB products count as multiple USB devices, for example, a device containing USB hub and headset can count as two USB devices. For more information, see the USB device documentation.

Cisco IP Phones 8861, 8865, and 8865NR

The following section describe the attributes of the Cisco IP Phones 8861, 8865, and 8865NR.

Phone Connections

Connect your phone to the corporate IP telephony network as shown in the following diagram.



1	DC adaptor port (DC48V).	7	Handset connection.
2	AC-to-DC power supply (optional).	8	Analog headset connection (optional).
3	AC power wall plug (optional).	9	USB port
4	Network port (10/100/1000 SW) connection. IEEE 802.3at power enabled.	10	Audio In/Out ports
5	Access port (10/100/1000 PC) connection.	11	USB port
6	Auxiliary port.		

Note Each USB port supports the connection of up to five supported and nonsupported devices. Each device connected to the phone is included in the maximum device count. For example, your phone can support five USB devices (such as three key expansion modules, one hub, and one other standard USB device) on the side port and five additional standard USB devices on the back port. Many third-party USB products count as multiple USB devices, for example, a device containing USB hub and headset can count as two USB devices. For more information, see the USB device documentation.

Buttons and Hardware

The Cisco IP Phone 8800 Series has two distinct hardware types:

- Cisco IP Phones 8811, 8841, 8851, 8851NR, and 8861—do not have a camera.
- Cisco IP Phones 8845, 8865, and 8865NR—have a built-in camera.

The following figure shows the Cisco IP Phone 8845.

Figure 1: Cisco IP Phone 8845 Buttons and Hardware



The following table describes the Cisco IP Phone 8800 Series Buttons.

Table 18: Cisco IP Phone 8800 Series Buttons

1	Handset and Handset light strip	Indicates whether you have an incoming call (flashing red) or a new voice message (steady red).
2	Camera Cisco IP Phone 8845, 8865, and 8865NR only	Use the camera for video calls.

3	Programmable feature buttons and line buttons	 Access your phone lines, features, and call sessions. When adding features to the phone line keys, you are limited by the number of line keys available. You cannot add more features than the number of line keys on your phone. For more information, see the Softkey, Line, and Feature Buttons section in the "Cisco IP Phone Hardware" chapter.
4	Softkey buttons	 Access to functions and services. For more information, see the Softkey, Line, and Feature Buttons section in the "Cisco IP Phone Hardware" chapter.
5	**Back**, Navigation cluster, and **Release**	**Back**  Return to the previous screen or menu. Navigation cluster  Navigation ring and **Select** button—Scroll through menus, highlight items and select the highlighted item. **Release**  End a connected call or session.
6	**Hold/Resume**, **Conference**, and **Transfer**	**Hold/Resume**  Place an active call on hold and resume the held call. **Conference**  Create a conference call. **Transfer**  Transfer a call.
7	**Speakerphone**, **Mute**, and **Headset**	**Speakerphone**  Toggle the speakerphone on or off. When the speakerphone is on, the button is lit. **Mute**  Toggle the microphone on or off. When the microphone is muted, the button is lit. **Headset**  Toggle the headset on. When the headset is on, the button islit. To leave headset mode, you pick up the handset or select **Speakerphone** .
8	**Contacts**, **Applications**, and **Messages**	**Contacts**  Access personal and corporate directories. **Applications**  Access recent calls, user preferences, phone settings, and phone model information. **Messages**  Autodial your voice messaging system.
9	**Volume** button	 Adjust the handset, headset, and speakerphone volume (off hook) and the ringer volume (on hook).

Softkey, Line, and Feature Buttons

You can interact with the features on your phone in several ways:

- Softkeys, located below the screen, give you access to the function displayed on the screen above the softkey. The softkeys change depending on what you are doing at the time. The **More ...** softkey shows you that more functions are available.
- Feature and line buttons, located on either side of the screen, give you access to phone features and phone lines.
  - Feature buttons—Used for features such as **Speed dial** or **Call pickup**, and to view your status on another line.
  - Line buttons—Used to answer a call or resume a held call. When not used for an active call, used to initiate phone functions, such as the missed calls display.

Feature and line buttons illuminate to indicate status.

LED Color and State	Normal Line Mode: Line Buttons	Normal Line Mode: Feature Buttons Enhanced Line Mode
Green, steady LED	Active call or two-way intercom call, held call, privacy in use	Active call or two-way intercom call, privacy in use
Green, flashing LED	Not applicable	Held call
Amber, steady LED	Incoming call, reverting call, one-way intercom call, logged into a Hunt Group	One-way intercom call, logged into a Hunt Group
Amber, flashing LED	Not applicable	Incoming call, reverting call
Red, steady LED	Remote line in use, Remote line on hold, Do Not Disturb active	Remote line in use, Do Not Disturb active
Red, flashing LED	Not applicable	Remote line on hold

Your administrator can set up some functions as softkeys or as feature buttons. You can also access some functions with softkeys or the associated hard button.

Protect Your Video Phone Camera

The camera on your video phone is fragile and could break during transportation of the phone.

Before you begin

You need one of these:

- Original phone box and the packing material
- Packaging material, such as foam or bubble wrap

Procedure

Step 1 If you have the original box:

a) Place the foam on the camera in such a way that the lens is well-protected.

b) Place the phone in its original box.

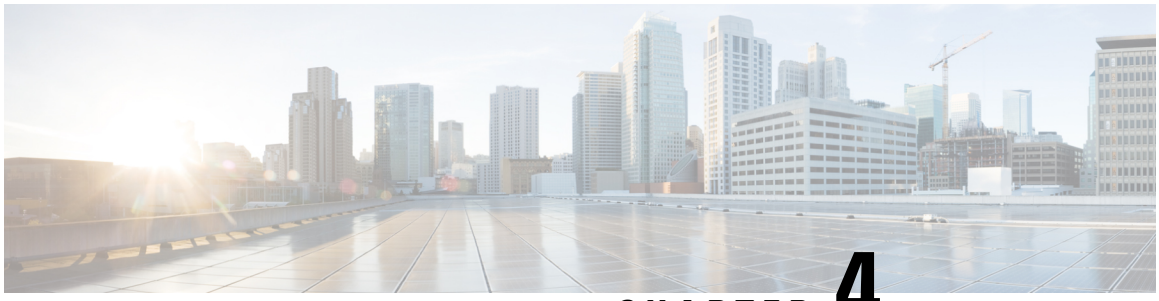
Step 2 If you do not have the box, carefully wrap the phone with foam or bubble wrap to protect the camera. Ensure that the foam protects and surrounds the camera so that nothing can press against the camera from any direction or the camera may be damaged in transport.



PART II

Cisco IP Phone Installation

- Cisco IP Phone Installation, on page 41
- Cisco Unified Communications Manager Phone Setup, on page 65
- Self Care Portal Management, on page 77



CHAPTER 4

Cisco IP Phone Installation

- Verify the Network Setup, on page 41
- Activation Code Onboarding for On-premises Phones, on page 42
- Activation Code Onboarding and Mobile and Remote Access, on page 43
- Enable Autoregistration for Phones, on page 43
- Install Cisco IP Phone, on page 45
- Set Up Phone from Setup Menus, on page 47
- Enable the Wireless LAN on the Phone, on page 49
- Configure Network Settings, on page 55
- Phone Startup Verification, on page 62
- Configure Phone Services for Users, on page 62
- Change a User's Phone Model, on page 63

Verify the Network Setup

As they deploy a new IP telephony system, system administrators and network administrators must complete several initial configuration tasks to prepare the network for IP telephony service. For information and a checklist for setting up and configuring a Cisco IP telephony network, see the documentation for your particular Cisco Unified Communications Manager release.

For the phone to operate successfully as an endpoint in your network, your network must meet specific requirements. One requirement is the appropriate bandwidth. The phones require more bandwidth than the recommended 32 kbps when they register to Cisco Unified Communications Manager. Consider this higher bandwidth requirement when you configure your QoS bandwidth. For more information, refer to *Cisco Collaboration System 12.x Solution Reference Network Designs (SRND)* or later (https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12.html).

Note The phone displays the date and time from Cisco Unified Communications Manager. The time displayed on the phone can differ from the Cisco Unified Communications Manager time by up to 10 seconds.

Procedure

Step 1 Configure a VoIP Network to meet the following requirements:

- VoIP is configured on your routers and gateways.
- Cisco Unified Communications Manager is installed in your network and is configured to handle call processing.

Step 2 Set up the network to support one of the following:

- DHCP support
- Manual assignment of IP address, gateway, and subnet mask

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Activation Code Onboarding for On-premises Phones

You can use Activation Code Onboarding to quickly set up new phones without autoregistration. With this approach, you control the phone onboarding process using the one of the following:

- Cisco Unified Communications Bulk Administration Tool (BAT)
- Cisco Unified Communications Manager Administration interface
- Administrative XML Web Service (AXL)

Enable this feature from the **Device Information** section of the Phone Configuration page. Select **Require Activation Code for Onboarding** if you want this feature to apply to a single on-premises phone.

Users must enter an activation code before their phones can register. Activation Code Onboarding can be applied to individual phones, a group of phones, or across an entire network.

This is an easy way for users to onboard their phones because they only enter a 16-digit activation code. Codes are entered either manually or with a QR code if a phone has a video camera. We recommend that you use a secure method to give users this information. But if a user is assigned to a phone, then this information is available on the Self Care Portal. The audit log records when a user accesses the code from the portal.

Activation codes can only be used once, and they expire after 1 week by default. If a code expires, you will have to provide the user with a new one.

You will find this approach an easy way to keep your network secure because a phone cannot register until the Manufacturing Installed Certificate (MIC) and activation code are verified. This method is also a convenient way to bulk onboard phones because it doesn't use the Tool for Auto-registered Phone Support (TAPS) or autoregistration. The rate of onboarding is one phone per second or about 3600 phones per hour. Phones can be added with the Cisco Unified Communications Manager Administrative, with Administrative XML Web Service (AXL), or with BAT.

Existing phones reset after they are configured for Activation Code Onboarding. They don't register until the activation code is entered and the phone MIC is verified. Inform current users that you are moving towards Activation Code Onboarding before you implement it.

For more information, see *Administration Guide for Cisco Unified Communications Manager and IM and Presence Service, Release 12.0(1)* or later.

Activation Code Onboarding and Mobile and Remote Access

You can use Activation Code Onboarding with Mobile and Remote Access when deploying Cisco IP phones for remote users. This feature is a secure way to deploy off-premises phones when autoregistration is not required. But you can configure a phone for autoregistration when on-premises, and activation codes when off-premises. This feature is similar to Activation Code Onboarding for on-premises phones, but it makes activation code available for off-premises phones also.

Activation Code Onboarding for Mobile and Remote Access requires Cisco Unified Communications Manager 12.5(1)SU1 or later, and Cisco Expressway X12.5 or later. Smart Licensing should be enabled also.

You enable this feature from the Cisco Unified Communications Manager Administration, but note the following:

- Enable this feature from the **Device Information** section of the Phone Configuration page.
- Select **Require Activation Code for Onboarding** if you want this feature to apply just to a single on-premises phone.
- Select **Allow Activation Code via MRA** and **Require Activation Code for Onboarding** if you want to use Activation Onboarding for a single off-premises phone. If the phone is on-premises, it changes to Mobile and Remote Access mode and uses the Expressway. If the phone cannot reach the Expressway, it does not register until it is off premises.

For more information, see the following documents:

- *Administration Guide for Cisco Unified Communications Manager and IM and Presence Service, Release 12.0(1)*
- *Mobile and Remote Access Through Cisco Expressway* for Cisco Expressway X12.5 or later

Enable Autoregistration for Phones

The Cisco IP Phone requires Cisco Unified Communications Manager to handle call processing. See the documentation for your particular Cisco Unified Communications Manager release or the context-sensitive help in the Cisco Unified Communications Manager Administration to ensure that Cisco Unified Communications Manager is set up properly to manage the phone and to properly route and process calls.

Before you install the Cisco IP Phone, you must choose a method for adding phones to the Cisco Unified Communications Manager database.

By enabling autoregistration before you install the phones, you can:

- Add phones without first gathering MAC addresses from the phones.
- Automatically add a Cisco IP Phone to the Cisco Unified Communications Manager database when you physically connect the phone to your IP telephony network. During autoregistration, Cisco Unified Communications Manager assigns the next available sequential directory number to the phone.
- Quickly enter phones into the Cisco Unified Communications Manager database and modify any settings, such as the directory numbers, from Cisco Unified Communications Manager.

- Move autoregistered phones to new locations and assign them to different device pools without affecting their directory numbers.

Autoregistration is disabled by default. In some cases, you might not want to use autoregistration; for example, if you want to assign a specific directory number to the phone, or if you want to use a secure connection with Cisco Unified Communications Manager. For information about enabling autoregistration, see the documentation for your particular Cisco Unified Communications Manager release. When you configure the cluster for mixed mode through the Cisco CTL client, autoregistration is automatically disabled, however you can enable it. When you configure the cluster for nonsecure mode through the Cisco CTL client, autoregistration is not enabled automatically.

You can add phones with autoregistration and TAPS, the Tool for AutoRegistered Phones Support, without first gathering MAC addresses from phones.

TAPS works with the Bulk Administration Tool (BAT) to update a batch of phones that were already added to the Cisco Unified Communications Manager database with dummy MAC addresses. Use TAPS to update MAC addresses and to download predefined configurations for phones.

Cisco recommends that you use autoregistration and TAPS to add fewer than 100 phones to your network. To add more than 100 phones to your network, use the Bulk Administration Tool (BAT).

To implement TAPS, you or the end user dials a TAPS directory number and follows voice prompts. After the process is complete, the phone contains the directory number and other settings, and the phone is updated in Cisco Unified Communications Manager Administration with the correct MAC address.

Verify that autoregistration is enabled and is properly configured in Cisco Unified Communications Manager Administration before you connect any Cisco IP Phone to the network. For information about enabling and configuring autoregistration, see the documentation for your particular Cisco Unified Communications Manager release.

Autoregistration must be enabled in Cisco Unified Communications Manager Administration for TAPS to function.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, click **System** > **Cisco Unified CM**.

Step 2 Click **Find** and select the required server.

Step 3 In **Auto-registration Information**, configure these fields.

- **Universal Device Template**
- **Universal Line Template**
- **Starting Directory Number**
- **Ending Directory Number**

Step 4 Uncheck the **Auto-registration Disabled on this Cisco Unified Communications Manager** check box.

Step 5 Click **Save**.

Step 6 Click **Apply Config**.

Install Cisco IP Phone

After the phone connects to the network, the phone startup process begins, and the phone registers with Cisco Unified Communications Manager. To finish installing the phone, configure the network settings on the phone depending on whether you enable or disable DHCP service.

If you used autoregistration, you need to update the specific configuration information for the phone such as associating the phone with a user, changing the button table, or directory number.

Note Before using external devices, read External Devices, on page 23.

For information about installing accessories, see *Cisco IP Phone 7800 and 8800 Series Accessories Guide for Cisco Unified Communications Manager*.

If you only have one LAN cable at your desk, you can plug your phone into the LAN with the SW port and then connect your computer into the PC port. For more information, see Share a Network Connection with Your Phone and Computer, on page 46.

You can also daisy chain two phones together. Connect the PC port of the first phone to the SW port of the second phone.

Caution Do not connect the SW and PC ports into the LAN.

Procedure

Step 1 Choose the power source for the phone:

- Power over Ethernet (PoE)
- External power supply

For more information, see Phone Power Requirements, on page 14.

Step 2 Connect the handset to the handset port and press the cable into the channel in the phone.

The wideband-capable handset is designed especially for use with a Cisco IP Phone. The handset includes a light strip that indicates incoming calls and waiting voice messages.

Caution Failure to press the cable into the channel in the phone can damage the printed circuit board. The cable channel reduces the strain on the connector and the printed circuit board.

Step 3 Connect a headset or wireless headset. You can add a headset later if you do not connect one now.

Press the cable into the cable channel.

Caution Failure to press the cable into the channel in the phone can damage the printed circuit board inside the phone. The cable channel reduces the strain on the connector and the printed circuit board.

Step 4 Connect a straight-through Ethernet cable from the switch to the network port labeled 10/100/1000 SW on the Cisco IP Phone. Each Cisco IP Phone ships with one Ethernet cable in the box.

Use Category 3, 5, 5e, or 6 cabling for 10 Mbps connections; Category 5, 5e, or 6 for 100Mbps connections; and Category 5e or 6 for 1000 Mbps connections. For more information, see Network and Computer Port Pinouts, on page 12 for guidelines.

Step 5 Connect a straight-through Ethernet cable from another network device, such as a desktop computer, to the computer port on the Cisco IP Phone. You can connect another network device later if you do not connect one now.

Use Category 3, 5, 5e, or 6 cabling for 10 Mbps connections; Category 5, 5e, or 6 for 100Mbps connections; and Category 5e or 6 for 1000 Mbps connections. For more information, see Network and Computer Port Pinouts, on page 12 for guidelines.

Step 6 If the phone is on a desk, adjust the footstand. With a wall-mounted phone, you might need to adjust the handset rest to ensure that the receiver cannot slip out of the cradle.

Step 7 Monitor the phone startup process. This step adds primary and secondary directory numbers and features that are associated with directory numbers to the phone, and verifies that the phone is configured properly.

Step 8 If you are configuring the network settings on the phone, you can set up an IP address for the phone by either using DHCP or manually entering an IP address.

See Configure Network Settings, on page 55 and Network Setup, on page 225.

Step 9 Upgrade the phone to the current firmware image.

Firmware upgrades over the WLAN interface may take longer than upgrading over the wired interface, depending on the quality and bandwidth of the wireless connection. Some upgrades may take more than one hour.

Step 10 Make calls with the Cisco IP Phone to verify that the phone and features work correctly.

See the *Cisco IP Phone 8800 Series User Guide*.

Step 11 Provide information to end users about how to use their phones and how to configure their phone options. This step ensures that users have adequate information to successfully use their Cisco IP Phones.

Share a Network Connection with Your Phone and Computer

Both your phone and your computer must connect to your network to function. If you only have one Ethernet port, then your devices can share the network connection.

Before you begin

Your administrator must enable the PC port in Cisco Unified Communications Manager before you can use it.

Procedure

Step 1 Connect the phone SW port to the LAN with an Ethernet cable.

Step 2 Connect your computer to the phone PC port with an Ethernet cable.

Set Up Phone from Setup Menus

The Cisco IP Phone includes the following configuration menus:

- Network Setup: Provides options for viewing and configuring network settings such as IPv4-only, IPv6-only, WLAN, and Ethernet.
- Ethernet Setup: The menu items in this submenu provide configuration options to configure the Cisco IP Phone over an ethernet network.
- WiFi Client Setup: The menu items in this submenu provide configuration options to configure the Cisco IP Phone with the wireless local area network (WLAN). Wi-Fi is only supported on Cisco IP Phone 8861 and 8865.

Note The phone PC port is disabled when Wi-Fi is enabled on your phone.

- IPv4 Setup and IPv6 Setup: These submenus of the Ethernet Setup menu and of the WiFi Client Setup menu provide additional network options.
- Security Setup: Provides options for viewing and configuring security settings such as security mode, the trust list and 802.1X authentication.

Before you can change option settings on the Network Setup menu, you must unlock options for editing.

Note You can control whether a phone has access to the Settings menu or to options on this menu by using the Settings Access field in the Cisco Unified Communications Manager Administration Phone Configuration window. The Settings Access field accepts these values:

- Enabled: Allows access to the Settings menu.
- Disabled: Prevents access to the Settings menu.
- Restricted: Allows access to the User Preferences menu and allows volume changes to be saved. Prevents access to other options on the Settings menu.

If you cannot access an option on the Administrator Settings menu, check the Settings Access field.

Procedure

Step 1 Press **Applications** .

Step 2 Select **Admin settings**.

Step 3 Select **Network setup** or **Security setup**.

Step 4 Enter your user ID and password, if required, and click **Sign-In**.

Step 5 Perform one of these actions to display the desired menu:

- Use the navigation arrows to select the desired menu and then press **Select**.
- Use the keypad on the phone to enter the number that corresponds to the menu.

Step 6 To display a submenu, repeat step 5.

Step 7 To exit a menu, press **Exit** or the back arrow .

Apply a Phone Password

You can apply a password to the phone. If you do, no changes can be made to the administrative options on the phone without password entry on the Admin Settings phone screen.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, navigate to the Common Phone Profile Configuration window (**Device** > **Device Settings** > **Common Phone Profile**).

Step 2 Enter a password in the Local Phone Unlock Password option.

Step 3 Apply the password to the common phone profile that the phone uses.

Text and Menu Entry from Phone

When you edit the value of an option setting, follow these guidelines:

- Use the arrows on the navigation pad to highlight the field that you wish to edit, then press **Select** in the navigation pad to activate the field. After the field is activated, you can enter values.
- Use the keys on the keypad to enter numbers and letters.
- To enter letters by using the keypad, use a corresponding number key. Press the key one or more times to display a particular letter. For example, press the **2** key once for "a," twice quickly for "b," and three times quickly for "c." After you pause, the cursor automatically advances to allow you to enter the next letter.
- Press the arrow softkey if you make a mistake. This softkey deletes the character to the left of the cursor.
- Press **Cancel** before pressing **Save** to discard any changes that you made.
- To enter an IP address, you enter values into four segments already divided for you. When you have finished entering the leftmost digits before the first period, use the right arrow key to move to the next segment. The period that follows the leftmost digits is automatically inserted.
- To enter a colon for an IPv6 address, press ***** on the keypad.



Note The Cisco IP Phone provides several methods to reset or restore option settings, if necessary.

Related Topics

Basic Reset, on page 259
Apply a Phone Password, on page 48

Enable the Wireless LAN on the Phone

Before you set up a wireless LAN, check to see that your phone supports wireless use. The Cisco IP Phone 8861 and 8865 support a wireless LAN deployment. The Cisco IP Phone 8865NR does not support a wireless LAN.

Ensure that the Wi-Fi coverage in the location where the wireless LAN is deployed is suitable for transmitting voice packets.

If you have enabled the Wi-Fi connectivity for voice and you're using EAP-FAST or PEAP security mode, authenticate the Wi-Fi network with the WLAN Sign in application. WEP, PSK, and open security modes authenticate on the Wi-Fi network.

A fast-secure roaming method is recommended for Wi-Fi users.

Note The phone PC port is disabled when Wi-Fi is enabled on your phone.

For complete configuration information, see the *Cisco IP Phone 8800 Wireless LAN Deployment Guide* at this location:

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

The *Cisco IP Phone 8800 Wireless LAN Deployment Guide* includes the following configuration information:

- Wireless network configuration
- Wireless network configuration in Cisco Unified Communications Manager Administration
- Wireless network configuration on the Cisco IP Phone

Before you begin

Make sure that Wi-Fi is enabled on the phone, and the Ethernet cable is disconnected.

Procedure

Step 1 To enable the application, press **Applications** .

Step 2 Go to **Admin settings** > **Network setup** > **Wi-Fi Client setup** > **Network name**.
You see a list of available wireless access points to which you can connect.

Step 3 Enable the Wireless network.

Set Up the Wireless LAN from Cisco Unified Communications Manager

In Cisco Unified Communications Manager Administration, you must enable a parameter called “Wi-Fi” for the wireless Cisco IP Phone.

Note In the Phone Configuration window in Cisco Unified Communications Manager Administration (**Device** > **Phone**), use the wired-line MAC address when you configure the MAC address. Cisco Unified Communications Manager registration does not use the wireless MAC address.

Perform the following procedure in Cisco Unified Communications Manager Administration.

Procedure

Step 1 To enable the wireless LAN on a specific phone, perform the following steps:

a) Select **Device** > **Phone**.
b) Locate the required phone.
c) Select the **Enabled** setting for the Wi-Fi parameter in the Product Specific Configuration Layout section.
d) Check the **Override Common Settings** check box.

Step 2 To enable wireless LAN for a group of phones,

a) Select **Device** > **Device Settings** > **Common Phone Profile**.
b) Select the **Enabled** setting for the Wi-Fi parameter.

Note To ensure that the configuration in this step works, uncheck the **Override Common Settings** check box mentioned in Step 1d.

c) Check the **Override Common Settings** check box.
d) Associate the phones with that common phone profile using **Device** > **Phone**.

Step 3 To enable wireless LAN for all WLAN-capable phones in your network,

a) Select **System** > **Enterprise Phone Configuration**.
b) Select the **Enabled** setting for the Wi-Fi parameter.

Note To ensure that the configuration in this step works, uncheck the **Override Common Settings** check box mentioned in Step 1d and Step 2c.

c) Check the **Override Common Settings** check box.

Set Up Wireless LAN from Phone

Before the Cisco IP Phone can connect to the WLAN, you must configure the network profile for the phone with the appropriate WLAN settings. You can use the **Network setup** menu on the phone to access the **Wi-Fi client setup** submenu and set up the WLAN configuration.

Note The phone PC port is disabled when Wi-Fi is enabled on your phone.

Note The **Wi-Fi client setup** option does not appear in the **Network setup** menu when Wi-Fi is disabled on the Cisco Unified Communications Manager.

For additional information, see *Cisco IP Phone 8800 Series WLAN Deployment Guide*, located here: http://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8800-series/products-implementation-design-guides-list.html.

The **User Modifiable** field in the wireless LAN profile controls that ability of the user to configure security modes on the phone. When a user cannot change some of the fields, the fields display in gray.

Before you begin

Configure the wireless LAN from Cisco Unified Communications Manager.

Procedure

Step 1 Press **Applications** .

Step 2 Select **Admin settings** > **Network setup** > **Wi-Fi client setup**.

Step 3 Set up the wireless configuration as described in the following table.

Table 19: WiFi Client Setup Menu Options

Option	Description	To change
Network name	Specifies the Service Set Identifier, a unique identifier for accessing wireless access points. Displays list of available wireless access points.	See Configure Network Settings,
IPv4-only Setup	In the IPv4 Setup configuration submenu, you can do the following: • Enable or disable the phone to use the IP address that the DHCP server assigns. • Manually set the IP Address, Subnet Mask, Default Routers, DNS Server, and Alternate TFTP servers. For more information about the IPv4 address fields, see IPv4 Fields, on page 57.	Scroll to IPv4 Setup and press **Sel**

Option	Description	To change
IPv6-only Setup	In the IPv6 Setup configuration submenu, you can do the following: • Enable or disable the phone to use the IPv6 address that is either assigned by DHCPv6 server or acquired by SLAAC through an IPv6-enabled router. • Manually set the IPv6 Address, Prefix Length, Default Routers, DNS Server, and Alternate TFTP servers. For more information about the IPv6 address fields, see IPv6 Fields, on page 59.	Scroll to IPv6 Setup and press **Select**.
MAC Address	Unique Media Access Control (MAC) address of the phone.	Display only. Cannot configure.
Domain Name	Name of the Domain Name System (DNS) domain in which the phone resides.	See Configure Network Settings, on p

Step 4 Press **Save** to make changes or press **Revert** to discard the connection.

Set the Number of WLAN Authentication Attempts

An authentication request is a confirmation of the user's sign-in credentials. It occurs whenever a phone that has already joined a Wi-Fi network tries to reconnect to the Wi-Fi server. Examples include when a Wi-Fi session times out or a Wi-Fi connection is lost and then reacquired.

You can configure the number of times a Wi-Fi phone sends an authentication request to the Wi-Fi server. The default number of attempts is 2, but you can set this parameter from 1 to 3. If a phone fails the authentication, then the user is prompted to sign in again.

You can apply WLAN Authentication Attempts to individual phones, to a pool of phones, or to all the Wi-Fi phones in your network.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **Device** > **Phone** and locate the phone.
Step 2 Navigate to the Product Specific Configuration area and set the **WLAN Authentication Attempts** field.
Step 3 Select **Save**.
Step 4 Select **Apply Config**.
Step 5 Restart the phone.

Enable WLAN Prompt Mode

Enable WLAN Profile 1 Prompt Mode if you want a user to sign into the Wi-Fi network when their phone powers-up or resets.

Procedure

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

Step 2 Locate the phone that you need to set up.

Step 3 Navigate to the Product Specific Configuration area and set the **WLAN Profile 1 Prompt Mode** field to **Enable**.

Step 4 Select **Save**.

Step 5 Select **Apply Config**.

Step 6 Restart the phone.

Set Up a Wi-Fi Profile using Cisco Unified Communications Manager

You can configure a Wi-Fi profile and then assign the profile to the phones that support Wi-Fi. The profile contains the parameters required for phones to connect to the Cisco Unified Communications Manager with Wi-Fi. When you create and use a Wi-Fi profile, you or your users do not need to configure the wireless network for individual phones.

Wi-Fi profiles are supported on Cisco Unified Communications Manager Release 10.5(2) or later. EAP-FAST, PEAP-GTC, and PEAP-MSCHAPv2 are supported in Cisco Unified Communications Manager Release 10.0 and later. EAP-TLS is supported in Cisco Unified Communications Manager Release 11.0 and later.

A Wi-Fi profile enables you to prevent or limit changes to the Wi-Fi configuration on the phone by the user.

We recommend that you use a secure profile with TFTP encryption enabled to protect keys and passwords when you use a Wi-Fi profile.

When you set up the phones to use EAP-FAST, PEAP-MSCHAPv2, or PEAP-GTC authentication, your users need individual user ids and passwords to sign into the phone.

The phones only support one server certificate which can be installed either with SCEP or the manual install method but not both methods. The phones don't support the TFTP method of certificate installation.

Note Phones that use Mobile and Remote Access through Expressway to connect to the Cisco Unified Communications Manager cannot use the Wi-Fi profile. Because you do not have the SSID, authentication mode, and login credentials of the user's phone, you cannot configure a wireless LAN profile for their phone.

Procedure

Step 1 In the Cisco Unified Communications Administration, select **Device** > **Device Settings** > **Wireless LAN Profile**.

Step 2 Click **Add New**.

Step 3 In the **Wireless LAN Profile Information** section, set the parameters:

- **Name**—Enter a unique name for the Wi-Fi profile. This name displays on the phone.
- **Description**—Enter a description for the Wi-Fi profile to help you differentiate this profile from other Wi-Fi profiles.
- **User Modifiable**—Select an option:
  - **Allowed**—Indicates that the user can make changes to the Wi-Fi settings from their phone. This option is selected by default.
  - **Disallowed**—Indicates that the user cannot make any changes to the Wi-Fi settings on their phone.
  - **Restricted**—Indicates that the user can change the Wi-Fi username and password on their phone. But users are not allowed to make changes to other Wi-Fi settings on the phone.

Step 4 In the **Wireless Settings** section, set the parameters:

- **SSID (Network Name)**—Enter the network name available in the user environment to which the phone can be connected. This name is displayed under the available network list on the phone and the phone can connect to this wireless network.
- **Frequency Band**—Available options are Auto, 2.4 GHz, and 5 GHz. This field determines the frequency band that the wireless connection uses. If you select Auto, the phone attempts to use the 5 GHz band first and only uses the 2.4 GHz band when the 5 GHz is not available.

Step 5 In the **Authentications Settings** section, set the **Authentication Method** to one of these authentication methods: EAP-FAST, EAP-TLS, PEAP-MSCHAPv2, PEAP-GTC, PSK, WEP, and None.

After you set this field, you may see extra fields that you need to set.

- **User certificate**—Required for EAP-TLS authentication. Select **Manufacturing installed** or **User installed**. The phone requires a certificate to be installed, either automatically from the SCEP or manually from the administration page on the phone.
- **PSK passphrase**—Required for PSK authentication. Enter the 8- 63 character ASCII or 64 HEX character pass phrase.
- **WEP Key**—Required for WEP authentication. Enter the 40/102 or 64/128 ASCII or HEX WEP key.
  - 40/104 ASCII is 5 characters.
  - 64/128 ASCII is 13 characters.
  - 40/104 HEX is 10 characters.
  - 64/128 HEX is 26 characters.
- **Provide Shared Credentials**: Required for EAP-FAST, PEAP-MSCHAPv2, and PEAP-GTC authentication.
  - If the user manages the username and password, leave the **Username** and **Password** fields blank.
  - If all your users share the same username and password, you can input the information in the **Username** and **Password** fields.

- Enter a description in the **Password Description** field.

Note If you need to assign each user a unique username and password, you need to create a profile for each user.

Note The **Network Access Profile** field is not supported by the Cisco IP Phone 8861 and 8865.

Step 6 Click **Save**.

What to do next

Apply the WLAN Profile Group to a device pool (**System** > **Device Pool**) or directly to the phone (**Device** > **Phone**).

Set Up a Wi-Fi Group using Cisco Unified Communications Manager

You can create a wireless LAN profile group and add any wireless LAN profile to this group. The profile group can then be assigned to the phone when you set up the phone.

Procedure

Step 1 In Cisco Unified Communications Administration, select **Device** > **Device Settings** > **Wireless LAN Profile Group**.

You can also define a wireless LAN profile group from **System** > **Device Pool**.

Step 2 Click **Add New**.

Step 3 In the **Wireless LAN Profile Group Information** section, enter a group name and description.

Step 4 In the **Profiles for this Wireless LAN Profile Group** section, select an available profile from the **Available Profiles** list and move the selected profile to the **Selected Profiles** list.

When more that one wireless LAN profile is selected, the phone uses only the first wireless LAN profile.

Step 5 Click **Save**.

Configure Network Settings

Procedure

Step 1 Press **Applications** .

Step 2 To access the Network Settings menu, select **Admin settings** > **Ethernet setup**.

Step 3 Set the fields as described in Ethernet Setup Fields, on page 56.

Step 4 After you have set the fields, select **Apply** and **Save**.

Step 5 Reboot the phone.

Ethernet Setup Fields

The Network Setup menu contains fields and submenus for IPv4 and IPv6. To change some of the fields, first disable DHCP.

Establishing a VPN connection overwrites the Ethernet data fields.

Table 20: Ethernet Setup Menu Options

Entry	Type	Description
IPv4 setup	Menu	See the IPv4 Fields section. This option displays only when the phone is configured in IPv4-only mode
IPv6 setup	Menu	See the “IPv6 Fields” section.
MAC Address	String	Unique Media Access Control (MAC) address of the phone. Display only. Cannot configure.
Domain Name	String	Name of the Domain Name System (DNS) domain in which the phone re To change this field, turn off DHCP.
Operational VLAN ID		Auxiliary Virtual Local Area Network (VLAN) configured on a Cisco Ca phone is a member. This setting is blank if the auxiliary VLAN or the Administrative VLAN If the phone has not received an auxiliary VLAN, this option indicates th The phone doesn't inherit the Operational VLAN from Admin VLAN if C or Link Level Discovery Protocol Media Endpoint Discovery is enabled. To assign a VLAN ID manually, use the Admin VLAN ID option.
Admin VLAN ID		Auxiliary VLAN of which the phone is a member. Used only if the phone does not receive an auxiliary VLAN from the swit ignored.
PC VLAN		Allows the phone to interoperate with third-party switches that do not sup Admin VLAN ID option must be set before you can change this option.

Entry	Type	Description
SW port setup	Auto Negotiate 1000 Full 100 Half 10 Half 10 Full	Speed and duplex of the network port. Valid values specify: • Auto Negotiate (default) • 1000 Full:1000-BaseT/full duplex • 100 Half: 100-BaseT/half duplex • 100 Full: 100-BaseT/full duplex • 10 Half: 10-BaseT/half duplex • 10 Full: 10-BaseT/full duplex If the phone is connected to a switch, configure the switch port to the configure both to autonegotiate. Unlock network configuration options if you want to edit this setting. this option, you must change the PC Port Configuration option to the
PC port setup	Auto Negotiate 1000 Full 100 Half 10 Half 10 Full	Speed and duplex of the Computer (access) port. Valid values: • Auto Negotiate (default) • 1000 Full: 1000-BaseT/full duplex • 100 Half: 100-BaseT/half duplex • 100 Full: 100-BaseT/full duplex • 10 Half: 10-BaseT/half duplex • 10 Full: 10-BaseT/full duplex If the phone is connected to a switch, configure the port on the switch t or configure both to autonegotiate. Unlock network configuration options if you want to change this field. must change the SW Port Configuration option to the same setting. To configure the setting on multiple phones simultaneously, enable R the Enterprise Phone Configuration window (**System** > **Enterprise P** If the ports are configured for Remote Port Configuration in Cisco Unif Administration, the data cannot be changed on the phone.

IPv4 Fields

Table 21: IPv4 Setup Menu Options

Entry	Description
DHCP Enabled	Indicates whether the phone has DHCP enabled or disabled. When DHCP is enabled, the DHCP server assigns the phone an IP address. When DHCP is disabled, the administrator must manually assign an IP address to the phone. For more information, see Set Up Phone to Use DHCP, on page 60 and Set Up Phone to Not Use DHCP, on page 61.

Entry	Description
IP Address	Internet Protocol (IP) address of the phone. If you assign an IP address with this option, you must also assign a subnet mask and default router. See the Subnet Mask and Default Router options in this table.
Subnet Mask	Subnet mask used by the phone.
Default Router	Default router used by the phone.
DNS Server 1 DNS Server 2 DNS Server 3	Primary Domain Name System (DNS) server (DNS Server 1) and optional backup DNS servers (DNS Server 2 and 3) that the phone uses.
Alternate TFTP	Indicates whether the phone is using an alternate TFTP server.
TFTP Server 1	Primary Trivial File Transfer Protocol (TFTP) server that the phone uses. If you are not using DHCP in your network and you want to change this server, you must use the TFTP Server 1 option. If you set the Alternate TFTP option to On, you must enter a nonzero value for the TFTP Server 1 option. If neither the primary TFTP server nor the backup TFTP server is listed in the CTL or ITL file on the phone, you must unlock the file before you can save changes to the TFTP Server 1 option. In this case, the phone deletes the file when you save changes to the TFTP Server 1 option. A new CTL or ITL file downloads from the new TFTP Server 1 address. When the phone looks for the TFTP server, the phone gives precedence to manually assigned TFTP servers, regardless of the protocol. If your configuration includes both IPv6 and IPv4 TFTP servers, the phone prioritizes the order that it looks for the TFTP server by giving priority to manually assigned IPv6 TFTP servers and IPv4 TFTP servers. The phone looks for the TFTP server in this order: **1.** Any manually assigned IPv4 TFTP servers **2.** Any manually assigned IPv6 servers **3.** DHCP assigned TFTP servers **4.** DHCPv6 assigned TFTP servers **Note** For information about the CTL and ITL files, see the *Cisco Unified Communications Manager Security Guide*.

Entry	Description
TFTP Server 2	Optional backup TFTP server that the phone uses if the primary TFTP server is unavailable. If neither the primary TFTP server nor the backup TFTP server is listed in the CTL or ITL file on the phone, you must unlock either of the files before you can save changes to the TFTP Server 2 option. In this case, the phone deletes either of the files when you save changes to the TFTP Server 2 option. A new CTL or ITL file downloads from the new TFTP Server 2 address. If you forget to unlock the CTL or ITL file, you can change the TFTP Server 2 address in either file, then erase them by pressing Erase from the Security Configuration menu. A new CTL or ITL file downloads from the new TFTP Server 2 address. When the phone looks for the TFTP server, it gives precedence to manually assigned TFTP servers, regardless of the protocol. If your configuration includes both IPv6 and IPv4 TFTP servers, the phone prioritizes the order that it looks for the TFTP server by giving priority to manually assigned IPv6 TFTP servers and IPv4 TFTP servers. The phone looks for the TFTP server in the following order: **1.** Any manually assigned IPv4 TFTP servers **2.** Any manually assigned IPv6 servers **3.** DHCP assigned TFTP servers **4.** DHCPv6 assigned TFTP servers **Note** For information about the CTL or ITL file, see Cisco Unified Communications Manager Security Guide.
BOOTP Server	Indicates whether the phone received the IP address from a BOOTP server rather than from a DHCP server.
DHCP Address Released	Releases the IP address that DHCP assigned. This field is editable if DHCP is enabled. If you wish to remove the phone from the VLAN and release the IP address for reassignment, set this option to Yes and press Apply.

IPv6 Fields

Before IPv6 setup options can be configured on your device, IPv6 must be enabled and configured in Cisco Unified Communication Administration. The following device configuration fields apply to IPv6 configuration:

- IP Addressing Mode
- IP Addressing Mode Preference for Signalling

If IPv6 is enabled in the Unified cluster, the default setting for IP addressing mode is IPv4 and IPv6. In this addressing mode, the phone acquires and uses one IPv4 address and one IPv6 address. It can use the IPv4 and the IPv6 address as required for media. The phone uses either the IPv4 or IPv6 address for call control signaling.

For more details about IPv6 deployment, see the IPv6 Deployment Guide for Cisco Collaboration Systems Release 12.0.

You set up IPv6 from one of the following menus:

- When Wi-Fi is disabled: **Ethernet Setup** > **IPv6 setup**
- When Wi-Fi is enabled: **Wi-Fi Client Setup** > **IPv6 setup**

Use the phone keypad to enter or edit an IPv6 address. To enter a colon, press the asterisk (*) on the keypad. To enter hexadecimal digits a, b, and c, press 2 on the keypad, scroll to select the required digit, and press **Enter**. To enter hexadecimal digits d, e, and f, press 3 on the keypad, scroll to select the required digit, and press **Enter**.

The following table describes the IPv6 related information found in the IPv6 menu.

Table 22: IPv6 Setup Menu Options

Entry	Default value	Description
DHCPv6 Enabled	Yes	Indicates the method that the phone use When DHCPv6 is enabled, the phone g the IPv6-enabled router. And if DHCPv stateless (from SLAAC) IPv6 address.
IPv6 Address	::	Displays the current IPv6-only address A valid IPv6 address is 128 bits in leng • Eight sets of hexadecimal digits se • Compressed format to collapse a s colon. If the IP address is assigned with this o
IPv6 Prefix Length	0	Displays the current prefix length for th The subnet prefix length is a decimal v
IPv6 Default Router	::	Displays the default router used by the
IPv6 DNS Server 1	::	Displays the primary DNSv6 server use
IPv6 DNS Server 2	::	Displays the secondary DNSv6 server u
IPv6 Alternate TFTP	No	Allows the user to enable the use of an
IPv6 TFTP Server 1	::	Displays the primary IPv6 TFTP server
IPv6 TFTP Server 2	::	(Optional) Displays the secondary IPv6 user to set a new secondary TFTP serve
IPv6 Address Released	No	Allows the user to release IPv6-related

Set Up Phone to Use DHCP

To enable DHCP and allow the DHCP server to automatically assign an IP address to the Cisco IP Phone and direct the phone to a TFTP server, perform these steps:

Procedure

Step 1 Press **Applications** .

Step 2 Choose **Admin settings** > **Network setup** > **Ethernet setup** > **IPv4 Setup**.

Step 3 To enable DHCP, set DHCP Enabled to **Yes**. DHCP is enabled by default.

Step 4 To use an alternate TFTP server, set Alternate TFTP Server to **Yes**, and enter the IP address for the TFTP Server.

Note Consult with the network administrator to determine whether you need to assign an alternative TFTP server instead of using the TFTP server that DHCP assigns.

Step 5 Press **Apply**,

Set Up Phone to Not Use DHCP

When not using DHCP, you must configure the IP address, subnet mask, TFTP server, and default router locally on the phone.

Procedure

Step 1 Press **Applications** .

Step 2 Choose **Admin settings** > **Network setup** > **Ethernet Setup** > **IPv4 Setup**.

Step 3 To disable DHCP and manually set an IP address:

a) Set DHCP Enabled to **No**.
b) Enter the static IP address for phone.
c) Enter the subnet mask.
d) Enter the default router IP addresses.
e) Set Alternate TFTP Server to **Yes**, and enter the IP address for TFTP Server 1.

Step 4 Press **Apply**.

Load Server

Load Server is used to optimize installation time for phone firmware upgrades and offload the WAN by storing images locally, which negates the need to traverse the WAN link for each phone upgrade.

You can set the Load Server to another TFTP server IP address or name (other than the TFTP Server 1 or TFTP Server 2) from which the phone firmware can be retrieved for phone upgrades. When the Load Server option is set, the phone contacts the designated server for the firmware upgrade.

Note A Load Server option allows you to specify an alternate TFTP server for phone upgrades only. The phone continues to use TFTP Server 1 or TFTP Server 2 to obtain configuration files. The Load Server option does not provide management of the process and of the files, such as file transfer, compression, or deletion.

The Load Server is configured from the Enterprise Phone Configuration window. From Cisco Unified Communications Manager Administration, choose **Device** > **Phone** >**Enterprise Phone Configuration**.

Phone Startup Verification

After the Cisco IP Phone has power connected to it, the phone begins the startup diagnostic process by cycling through the following steps.

1. The Feature and Session buttons flash amber and then green in sequence during the various stages of bootup as the phone checks the hardware.
2. The main screen displays `Registering to Cisco Unified Communications Manager`.

If the phone completes these stages successfully, it has started up properly and the **Select** button stays lit until it is selected.

Configure Phone Services for Users

You can give users access to Cisco IP Phone Services on the IP phone. You can also assign a button to different phone services. These services comprise XML applications and Cisco-signed Java midlets that enable the display of interactive content with text and graphics on the phone. The IP phone manages each service as a separate application. Examples of services include local movie times, stock quotes, and weather reports.

Before a user can access any service:

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

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

Before you set up services, gather the URLs for the sites that you want to set up and verify that users can access those sites from your corporate IP telephony network. This activity is not applicable for the default services that Cisco provides.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **Device** > **Device Settings** > **Phone Services**

Step 2 Verify that your users can access the Cisco Unified Communications Self Care Portal, from which they can select and subscribe to configured services.

See Self Care Portal Management, on page 77 for a summary of the information that you must provide to end users.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Change a User's Phone Model

You or your user can change a user's phone model. The change can be required for a number of reasons, for example:

- You have updated your Cisco Unified Communications Manager (Unified CM) to a software version that doesn't support the phone model.
- The user wants a different phone model from their current model.
- The phone requires repair or replacement.

The Unified CM identifies the old phone and uses the old phone's MAC address to identify the old phone configuration. The Unified CM copies the old phone configuration into the entry for the new phone. The new phone then has the same configuration as the old phone.

If you change an old phone with SCCP firmware to a model in the Cisco IP Phone 8800 Series, the new phone is configured for Session Line Mode.

If the old phone has a key expansion model configured, the Unified CM copies the expansion module information to the new phone at the same time. When the user connects a compatible key expansion module to the new phone, the new expansion module gets the migrated expansion module information.

If the old phone has a key expansion model configured and the new phone doesn't support an expansion module, the Unified CM doesn't copy the expansion module information.

Limitation: If the old phone has more lines or line buttons than the new phone, the new phone doesn't have the extra lines or line buttons configured.

The phone reboots when the configuration is complete.

Before you begin

Set up your Cisco Unified Communications Manager according to the instructions in the *Feature Configuration Guide for Cisco Unified Communications Manager*.

You need a new, unused phone that comes preinstalled with Firmware Release 12.8(1) or later.

Procedure

Step 1 Power off the old phone.

Step 2 Power on the new phone.

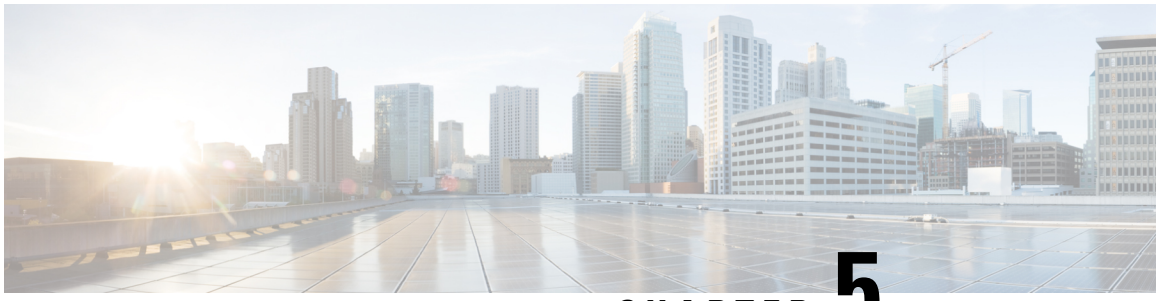
Step 3 On the new phone, select **Replace an existing phone**.

Step 4 Enter the primary extension of the old phone.

Step 5 If the old phone had a PIN assigned, enter the PIN.

Step 6 Press **Submit**.

Step 7 If there is more than one device for the user, select the device to replace and press **Continue**.



CHAPTER 5

Cisco Unified Communications Manager Phone Setup

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- Phone Addition Methods, on page 68
- Add Users to Cisco Unified Communications Manager, on page 70
- Add a User to an End User Group, on page 71
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- Survivable Remote Site Telephony, on page 72
- Enhanced Survivable Remote Site Telephony, on page 75
- Application Dial Rules, on page 75

Set Up Cisco IP Phone

If autoregistration is not enabled and the phone does not exist in the Cisco Unified Communications Manager database, you must configure the Cisco IP Phone in Cisco Unified Communications Manager manually. Some tasks in this procedure are optional, depending on your system and user needs.

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

Perform the configuration steps in the following procedure using Cisco Unified Communications Manager Administration.

Procedure

Step 1 Gather the following information about the phone:

- Phone model
- MAC address
- Physical location of the phone
- Name or user ID of phone user
- Device pool

- Partition, calling search space, and location information
- Number of lines and associated directory numbers (DNs) to assign to the phone
- Cisco Unified Communications Manager user to associate with the phone
- Phone usage information that affects phone button template, phone features, IP Phone services, or phone applications

The information provides a list of configuration requirements for setting up phones and identifies preliminary configuration that you need to perform before configuring individual phones, such as phone button templates.

Step 2 Verify that you have sufficient unit licenses for your phone.

Step 3 Customize phone button templates (if required) by changing the number of line buttons, speed-dial buttons or service URL buttons. Select **Device** > **Device Settings** > **Phone Button Template** to create and update the templates.

You can add a Privacy, All Calls, or Mobility button to meet user needs.

For more information, see Phone Button Templates, on page 185.

Step 4 Define the Device Pools. Select **System** > **Device Pool**.

Device Pools define common characteristics for devices, such as region, date/time group, softkey template, and MLPP information.

Step 5 Define the Common Phone Profile. Select **Device** > **Device settings** > **Common Phone Profile**.

Common phone profiles provide data that the Cisco TFTP server requires, as well as common phone settings, such as Do Not Disturb and feature control options.

Step 6 Define a Calling Search Space. In Cisco Unified Communications Manager Administration, click **Call Routing** > **Class of Control** > **Calling Search Space**.

A Calling Search Space is a collection of partitions that are searched to determine how a dialed number is routed. The calling search space for the device and the calling search space for the directory number are used together. The directory number CSS takes precedence over the device CSS.

Step 7 Configure a security profile for the device type and protocol. Select **System** > **Security** > **Phone Security Profile**.

Step 8 Add and configure the phone by completing the required fields in the Phone Configuration window. An asterisk (*) next to the field name indicates a required field; for example, MAC address and device pool.

This step adds the device with the default settings to the Cisco Unified Communications Manager database.

For information about product-specific configuration fields, see the “?” Button Help in the Phone Configuration window.

Note If you want to add both the phone and user to the Cisco Unified Communications Manager database at the same time, see the documentation for your particular Cisco Unified Communications Manager release.

Step 9 Add and configure directory numbers (lines) on the phone by completing the required fields in the Directory Number Configuration window. An asterisk (*) next to the field name indicates a required field; for example, directory number and presence group.

This step adds primary and secondary directory numbers and features associated with directory numbers to the phone.

Note If you do not configure the primary directory number, the user sees the message `Unprovisioned` on the phone.

Step 10 Configure speed-dial buttons and assign speed-dial numbers.

Users can change speed-dial settings on their phones by using Cisco Unified Communications Self Care Portal.

Step 11 Configure Cisco Unified IP Phone services and assign services (optional) to provide IP Phone services.

Users can add or change services on their phones by using the Cisco Unified Communications Self Care Portal.

Note Users can subscribe to the IP Phone service only if the Enterprise Subscription check box is unchecked when the IP Phone service is first configured in Cisco Unified Communications Manager Administration.

Note Some Cisco-provided default services are classified as enterprise subscriptions, so the user cannot add them through the Self Care Portal. Such services are on the phone by default, and they can only be removed from the phone if you disable them in Cisco Unified Communications Manager Administration.

Step 12 Assign services to programmable buttons (optional) to provide access to an IP Phone service or URL.

Step 13 Add user information by configuring required fields. An asterisk (*) next to the field name indicates a required field; for example, User ID and last name. This step adds user information to the global directory for Cisco Unified Communications Manager.

Note Assign a password (for Self Care Portal) and PIN (for Cisco Extension Mobility and Personal Directory).

Note If your company uses a Lightweight Directory Access Protocol (LDAP) directory to store information about users, you can install and configure Cisco Unified Communications to use your existing LDAP directory.

Note If you want to add both the phone and user to the Cisco Unified Communications Manager database at the same time, see the documentation for your particular Cisco Unified Communications Manager release.

Step 14 Associate a user to a user group. This step assigns users a common list of roles and permissions that apply to all users in a user group. Administrators can manage user groups, roles, and permissions to control the level of access (and, therefore, the level of security) for system users. For example, you must add users to the standard Cisco CCM End Users group so users can access Cisco Unified Communications Manager Self Care Portal.

Step 15 Associate a user with a phone (optional). This step provides users with control over their phone such a forwarding calls or adding speed-dial numbers or services.

Some phones, such as those in conference rooms, do not have an associated user.

Step 16 If you are not already in the End User Configuration window, choose **User Management** > **End User** to perform some final configuration tasks. Use the Search fields and **Find** to locate the user (for example, John Doe), then click on the user ID to get to the End User Configuration window for the user.

Step 17 In the Directory Number Associations area of the screen, set the primary extension from the drop-down list.

Step 18 In the Mobility Information area, check the Enable Mobility box.

Step 19 In the Permissions Information area, use the User Group buttons to add this user to any user groups.

For example, you may want to add the user to a group that is defined as a Standard CCM End User Group.

Step 20 To view all configured user groups, choose **User Management** > **User Group**.

Step 21 In the Extension Mobility area, check the Enable Extension Mobility Cross Cluster box if the user is allowed for Extension Mobility Cross Cluster service.

Step 22 Select **Save**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Determine the Phone MAC Address

To add phones to Cisco Unified Communications Manager, you must determine the MAC address of a phone.

Procedure

Perform one of the following actions:

- On the phone, press **Applications** , select **Phone Information** and look at the MAC Address field.
- Look at the MAC label on the back of the phone.
- Display the web page for the phone and click **Device Information**.

Phone Addition Methods

After you install the Cisco IP Phone, you can choose one of the following options to add phones to the Cisco Unified Communications Manager database.

- Add phones individually with Cisco Unified Communications Manager Administration
- Add multiple phones with the Bulk Administration Tool (BAT)
- Autoregistration
- BAT and the Tool for Auto-Registered Phones Support (TAPS)

Before you add phones individually or with BAT, you need the MAC address of the phone. For more information, see Determine the Phone MAC Address, on page 68.

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

Add Phones Individually

Collect the MAC address and phone information for the phone that you will add to the Cisco Unified Communications Manager.

Procedure

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

Step 2 Click **Add New**.

Step 3 Select the phone type.

Step 4 Select **Next**.

Step 5 Complete the information about the phone including the MAC Address.

For complete instructions and conceptual information about Cisco Unified Communications Manager, see the documentation for your particular Cisco Unified Communications Manager release.

Step 6 Select **Save**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Add Phones with a BAT Phone Template

The Cisco Unified Communications Bulk Administration Tool (BAT) enables you to perform batch operations, including registration of multiple phones.

To add phones using BAT only (not in conjunction with TAPS), you must obtain the appropriate MAC address for each phone.

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

Procedure

Step 1 From Cisco Unified Communications Administration, choose **Bulk Administration** > **Phones** > **Phone Template**.

Step 2 Click **Add New**.

Step 3 Choose a Phone Type and click **Next**.

Step 4 Enter the details of phone-specific parameters, such as Device Pool, Phone Button Template, and Device Security Profile.

Step 5 Click **Save**.

Step 6 Select **Device** > **Phone** > **Add New** to add a phone using the BAT phone template.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Add Users to Cisco Unified Communications Manager

You can display and maintain information about the users registered in Cisco Unified Communications Manager. Cisco Unified Communications Manager also allows each user to perform these tasks:

- Access the corporate directory and other customized directories from a Cisco IP Phone.
- Create a personal directory.
- Set up speed dial and call forwarding numbers.
- Subscribe to services that are accessible from a Cisco IP Phone.

Procedure

Step 1 To add users individually, see Add a User Directly to Cisco Unified Communications Manager, on page 71.

Step 2 To add users in batches, use the Bulk Administration Tool. This method also enables you to set an identical default password for all users.

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

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Add a User from an External LDAP Directory

If you added a user to an LDAP Directory (a non-Cisco Unified Communications Server directory), you can immediately synchronize the LDAP directory to the Cisco Unified Communications Manager on which you are adding the user and the user phone.

Note If you do not synchronize the LDAP Directory to the Cisco Unified Communications Manager immediately, the LDAP Directory Synchronization Schedule on the LDAP Directory window determines when the next autosynchronization is scheduled. Synchronization must occur before you can associate a new user to a device.

Procedure

Step 1 Sign into Cisco Unified Communications Manager Administration.

Step 2 Select **System** > **LDAP** > **LDAP Directory**.

Step 3 Use **Find** to locate your LDAP directory.

Step 4 Click on the LDAP directory name.

Step 5 Click **Perform Full Sync Now**.

Add a User Directly to Cisco Unified Communications Manager

If you are not using a Lightweight Directory Access Protocol (LDAP) directory, you can add a user directly with Cisco Unified Communications Manager Administration by following these steps.

Note If LDAP is synchronized, you cannot add a user with Cisco Unified Communications Manager Administration.

Procedure

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

Step 2 Click **Add New**.

Step 3 In the User Information pane, enter the following:

- User ID: Enter the end user identification name. Cisco Unified Communications Manager does not permit modifying the user ID after it is created. You may use the following special characters: =, +, <, >, #,;, \,, “”, and blank spaces. **Example**: johndoe
- Password and Confirm Password: Enter five or more alphanumeric or special characters for the end user password. You may use the following special characters: =, +, <, >, #, ;, \, , “”, and blank spaces.
- Last Name: Enter the end user last name. You may use the following special characters: =, +, <, >, #, ;, \, , “”, and blank spaces. **Example**: doe
- Telephone Number: Enter the primary directory number for the end user. End users can have multiple lines on their phones. **Example**: 26640 (John Doe’s internal company telephone number)

Step 4 Click **Save**.

Add a User to an End User Group

To add a user to the Cisco Unified Communications Manager Standard End User group, perform these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **User Management** > **User Settings** > **Access Control Group**.

The Find and List Users window displays.

Step 2 Enter the appropriate search criteria and click **Find**.

Step 3 Select the **Standard CCM End Users** link. The User Group Configuration window for the Standard CCM End Users appears.

Step 4 Select **Add End Users to Group**. The Find and List Users window appears.

Step 5 Use the Find User drop-down list boxes to find the users that you want to add and click **Find**.

A list of users that matches your search criteria appears.

Step 6 In the list of records that appear, click the check box next to the users that you want to add to this user group. If the list is long, use the links at the bottom to see more results.

Note The list of search results does not display users that already belong to the user group.

Step 7 Choose **Add Selected**.

Associate Phones with Users

You associate phones with users from the Cisco Unified Communications Manager End User window.

Procedure

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

The Find and List Users window appears.

Step 2 Enter the appropriate search criteria and click **Find**.

Step 3 In the list of records that appear, select the link for the user.

Step 4 Select **Device Association**.

The User Device Association window appears.

Step 5 Enter the appropriate search criteria and click **Find**.

Step 6 Choose the device that you want to associate with the user by checking the box to the left of the device.

Step 7 Choose **Save Selected/Changes** to associate the device with the user.

Step 8 From the Related Links drop-down list in the upper, right corner of the window, select **Back to User**, and click **Go**.

The End User Configuration window appears and the associated devices that you chose display in the Controlled Devices pane.

Step 9 Choose **Save Selected/Changes**.

Survivable Remote Site Telephony

Survivable Remote Site Telephony (SRST) ensures that basic phone functions remain accessible when WAN connectivity is lost. In this scenario, the phone can keep an in-progress call active, and the user can access a subset of the features available. When failover occurs, the user receives an alert message on the phone.

For more information about supported firmware and Survivable Remote Site Telephony, see *Cisco Unified Survivable Remote Site Telephony Compatibility Information* page on Cisco.com (http://www.cisco.com/c/en/us/support/unified-communications/unified-survivable-remote-site-telephony/products-device-support-tables-list.html).

The following table describes the availability of features during failover.

Table 23: SRST feature support

Feature	Supported	Notes
New Call	Yes	
End Call	Yes	
Redial	Yes	
Answer	Yes	
Hold	Yes	
Resume	Yes	
Conference	Yes	
Conference to Active Calls (Join)	No	The Active Calls softkey does not display.
Conference List	No	
Transfer	Yes	
Transfer to Active Calls (Direct Transfer)	No	
Auto Answer	Yes	
Call Waiting	Yes	
Caller ID	Yes	
Audible Message Waiting Indicator	Yes	
All Calls Programmable Line Key	Yes	
Answer Programmable Line Key	Yes	
Unified Session Presentation	Yes	Conference is the only feature supported due to other feature limitations.
Voicemail	Yes	Voicemail will not be synchronized with other users in the Cisco Unified Communications Manager cluster.

Feature	Supported	Notes
Call Forward All	Yes	Forward state is only available on the phone that sets the forward because there are no shared line appearances in SRST mode. The Call Forward All settings are not preserved on failover to SRST from the Cisco Unified Communications Manager, or from SRST fail-back to the Communications Manager. Any original Call Forward All still active on the Communications Manager should be indicated when the device reconnects to the Communications Manager after failover.
Speed Dial	Yes	
Service IRL Programmable Line Key	Yes	
To Voicemail (iDivert)	No	The iDivert softkey does not display.
Line Filters	Partial	Lines are supported but cannot be shared.
Park Monitoring	No	The Park softkey does not display.
Barge	No	The Barge softkey does not display.
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 does not display.
BLF	Partial	BLF feature key works like Speed Dial keys.
Hold Reversion	No	Calls remain on hold indefinitely.
Remote Hold	No	Calls appear as Local Hold calls.
Meet Me	No	The Meet Me softkey does not display.
PickUp	No	The softkey causes no action.
Group PickUp	No	The softkey causes no action.
Other PickUp	No	The softkey causes no action.
Malicious Call ID	No	The softkey causes no action.
QRT	No	The softkey causes no action.
Hunt Group	No	The softkey causes no action.
Intercom	No	The softkey causes no action.

Feature	Supported	Notes
Mobility	No	The softkey causes no action.
Privacy	No	The softkey causes no action.
Call Back	No	The Call Back softkey does not display.
Video	Yes	Video conference is not supported.
Video	Yes	Video conference is not supported.
Shared Line	No	
BLF Speed Dial	Yes	

Enhanced Survivable Remote Site Telephony

Enhanced Survivable Remote Site Telephony (E-SRST) ensures that additional phone features are available remain accessible when WAN connectivity is lost. In addition to the features supported by Survivable Remote Site Telephony(SRST), E-SRST supports the following:

- Shared-Line
- Busy-Lamp-Field (BLF)
- Video Calls

For more information about supported firmware and Survivable Remote Site Telephony, see *Cisco Unified Survivable Remote Site Telephony Compatibility Information* page on Cisco.com (http://www.cisco.com/c/en/us/support/unified-communications/unified-survivable-remote-site-telephony/products-device-support-tables-list.html).

Application Dial Rules

Application Dial Rules are used to convert numbers for shared mobile contacts to network dialable numbers. Application Dial Rules do not apply when the user is dialing a number manually, or if the number is edited before the user places the call.

Application Dial Rules are set in Cisco Unified Communications Manager.

For additional information about dial rules, see *System Configuration Guide for Cisco Unified Communications Manager*, “Configure Dial Rules” chapter.

Configure Application Dial Rules

Procedure

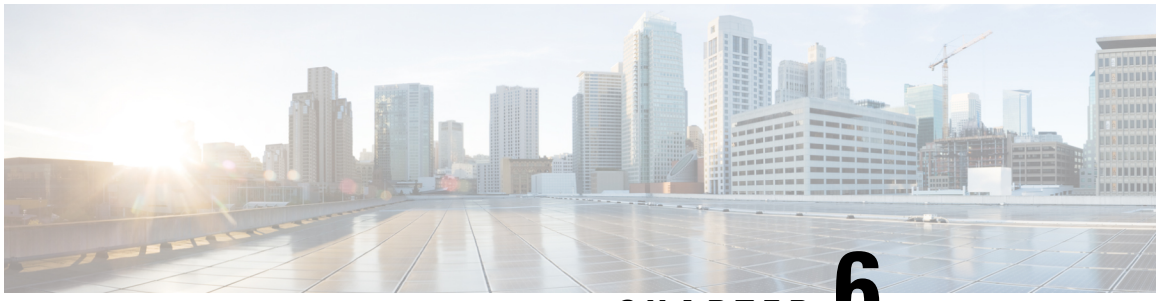
Step 1 In Cisco Unified Communications Manager Administration, go to **Call Routing** > **Dial Rules** > **Application Dial Rules**.

Step 2 Choose **Add New** to create a new application dial rule, or choose an existing application dial rule to edit it.

Step 3 Fill in the following fields:

- **Name** This field comprises a unique name for the dial rule that can contain up to 20 alphanumeric characters and any combination of spaces, periods (.), hyphens (-), and underscore characters (_).
- **Description** This field comprises a brief description that you enter for the dial rule.
- **Number Begins With** This field comprises the initial digits of the directory numbers to which you want to apply this application dial rule.
- **Number of Digits** This required field comprises the initial digits of the directory numbers to which you want to apply this application dial rule.
- **Total Digits to be Removed** This required field comprises the number of digits that you want Cisco Unified Communications Manager to remove from directory numbers that apply to this dial rule.
- **Prefix With Pattern** This required field comprises the pattern to prepend to directory numbers that apply to this application dial rule.
- **Application Dial Rule Priority** This field displays when you enter the Prefix With Pattern information. The field allows you to set the priority order of the application dial rules.

Step 4 Restart Cisco Unified Communications Manager.



CHAPTER 6

Self Care Portal Management

- Self Care Portal Overview, on page 77
- Set Up User Access to the Self Care Portal, on page 77
- Customize the Self Care Portal Display, on page 78

Self Care Portal Overview

From the Cisco Unified Communications Self Care Portal, users can customize and control phone features and settings.

As the administrator, you control access to the Self Care Portal. You must also provide information to your users so that they can access the Self Care Portal.

Before a user can access the Cisco Unified Communications Self Care Portal, you must use Cisco Unified Communications Manager Administration to add the user to a standard Cisco Unified Communications Manager End User group.

You must provide end users with the following information about the Self Care Portal:

- The URL to access the application. This URL is:

  `https://<server_name:portnumber>/ucmuser/`, where server_name is the host on which the web server is installed, and portnumber is the port number on that host.

- A user ID and default password to access the application.

- An overview of the tasks that users can accomplish with the portal.

These settings correspond to the values that you entered when you added the user to Cisco Unified Communications Manager.

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

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Set Up User Access to the Self Care Portal

Before a user can access the Self Care Portal, you need to authorize the access.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **User Management** > **End User**.
Step 2 Search for the user.
Step 3 Click the user ID link.
Step 4 Ensure that the user has a password and PIN configured.
Step 5 In the Permission Information section, ensure that the Groups list includes **Standard CCM End Users**.
Step 6 Select **Save**.

Customize the Self Care Portal Display

Most options display on the Self Care Portal. However, you must set the following options by using Enterprise Parameters Configuration settings in Cisco Unified Communications Manager Administration:

- Show Ring Settings
- Show Line Label Settings

Note The settings apply to all Self Care Portal pages at your site.

Procedure

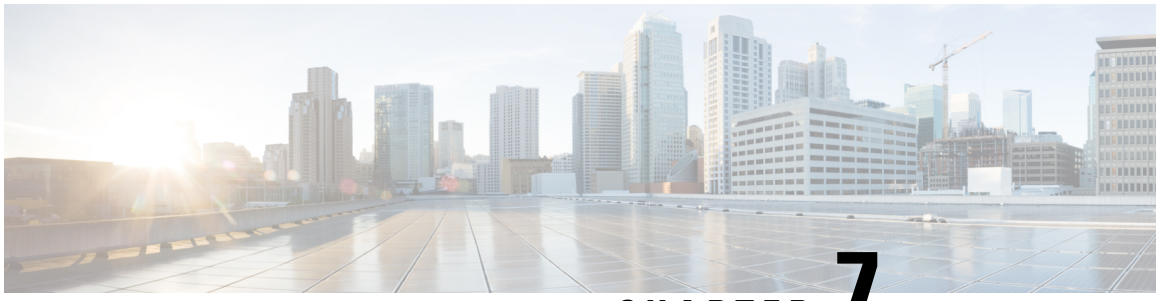
Step 1 In Cisco Unified Communications Manager Administration, select **System** > **Enterprise Parameters**.
Step 2 In the Self Care Portal area, set the **Self Care Portal Default Server** field.
Step 3 Enable or disable the parameters that the users can access in the portal.
Step 4 Select **Save**.



PART III

Cisco IP Phone Administration

- Cisco IP Phone Security, on page 81
- Cisco IP Phone Customization, on page 109
- Phone Features and Setup, on page 115
- Corporate and Personal Directory, on page 201



CHAPTER 7

Cisco IP Phone Security

- Security Enhancements for Your Phone Network, on page 81
- Supported Security Features, on page 82

Security Enhancements for Your Phone Network

You can enable Cisco Unified Communications Manager 11.5(1) or later version to operate in an enhanced security environment. With these enhancements, your phone network operates under a set of strict security and risk management controls to protect you and your users.

The enhanced security environment includes the following features:

- Contact search authentication.
- TCP as the default protocol for remote audit logging.
- FIPS mode.
- An improved credentials policy.
- Support for the SHA-2 family of hashes for digital signatures.
- Support for a RSA key size of 512 and 4096 bits.

Note Your Cisco IP Phone can only store a limited number of Identity Trust List (ITL) files. ITL files cannot exceed 64K limit on phone so limit the number of files that the Cisco Unified Communications Manager sends to the phone.

SIP OAuth Support

SIP OAuth mode allows you to use OAuth refresh tokens for phone authentication.

Cisco Unified Communications Manager (Unified CM) verifies the token presented by the phone and serves the configuration files only to authorized ones. OAuth token validation during SIP registration is completed when OAuth-based authorization is enabled on Unified CM cluster and Cisco IP phones.

Cisco IP phones support SIP OAuth authentication on Proxy Trivial File Transfer Protocol (TFTP) and Cisco Unified Survivable Remote Site Telephony (SRST).

- SIP OAuth on TFTP requirements:
  - Cisco Unified Communications Manager Release 14.0(1)SU1 or later
  - Cisco IP Phone Firmware Release 14.1(1) or later

Note Proxy TFTP and OAuth for Proxy TFTP aren't supported on Mobile Remote Access (MRA).

- SIP OAuth on SRST requirements:
  - Cisco Unified Communications Manager 14.0(1)SU1 or later
  - Cisco IP Phone Firmware Release 14.2(1) or later
  - Cisco SRST Software Release: IOS XE 17.8.1a or later
  - Cisco SRST Hardware Models: ISR1100, ISR43xx, ISR44xx, Catalyst 8200, or Catalyst 8300 platform

For information about how to configure SIP OAuth, see **SIP OAuth Mode** in *Security Guide for Cisco Unified Communications Manager*.

Where to Find More Information about Phone Security

For additional information about security, see the following:

- *Security Guide for Cisco Unified Communications Manager* (https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/14SU2/cucm_b_security-guide-14su2.html)
- *Cisco Unified SCCP and SIP SRST System Administration Guide* (https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_roadmap.html)
- *System Configuration Guide for Cisco Unified Communications Manager*, Release 14.0(1) or later (https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html).
- *Cisco IP Phone 7800 and 8800 Series Security Overview* (https://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-8800-series/white-paper-listing.html)

Supported Security Features

Security features protect against several threats, including threats to the identity of the phone and to data. These features establish and maintain authenticated communication streams between the phone and the Cisco Unified Communications Manager server, and ensure that the phone uses only digitally signed files.

Cisco Unified Communications Manager Release 8.5(1) and later includes Security by Default, which provides the following security features for Cisco IP Phones without running the CTL client:

- Signing of the phone configuration files

- Phone configuration file encryption
- HTTPS with Tomcat and other Web services

Note Secure signaling and media features still require you to run the CTL client and use hardware eTokens.

Implementing security in the Cisco Unified Communications Manager system prevents identity theft of the phone and Cisco Unified Communications Manager server, prevents data tampering, and prevents call signaling and media stream tampering.

To alleviate these threats, the Cisco IP telephony network establishes and maintains secure (encrypted) communication streams between a phone and the server, digitally signs files before they are transferred to a phone, and encrypts media streams and call signaling between Cisco IP Phones.

A Locally Significant Certificate (LSC) installs on phones after you perform the necessary tasks that are associated with the Certificate Authority Proxy Function (CAPF). You can use Cisco Unified Communications Manager Administration to configure an LSC, as described in the Cisco Unified Communications Manager Security Guide. Alternatively, you can initiate the installation of an LSC from the Security Setup menu on the phone. This menu also lets you update or remove an LSC.

A LSC cannot be used as the user certificate for EAP-TLS with WLAN authentication.

The phones use the phone security profile, which defines whether the device is nonsecure or secure. For information about applying the security profile to the phone, see the documentation for your particular Cisco Unified Communications Manager release.

If you configure security-related settings in Cisco Unified Communications Manager Administration, the phone configuration file contains sensitive information. To ensure the privacy of a configuration file, you must configure it for encryption. For detailed information, see the documentation for your particular Cisco Unified Communications Manager release.

The Cisco IP Phone 8800 Series complies with Federal Information Processing Standard (FIPS). To function correctly, FIPS mode requires a key size of 2048 bits or greater. If the certificate is not 2048 bits or greater, the phone will not register with Cisco Unified Communications Manager and `Phone failed to register. Cert key size is not FIPS compliant` displays on the phone.

If the phone has an LSC, you need to update the LSC key size to 2048 bits or greater before enabling FIPS.

The following table provides an overview of the security features that the phones support. For more information, see the documentation for your particular Cisco Unified Communications Manager release.

For view the current security settings on a phone, including Security mode, Trust list, and 802.1X Authentication, press **Applications** and choose **Admin Settings** > **Security setup**.

Table 24: Overview of Security Features

Feature	Description
Image authentication	Signed binary files (with the extension.sbn) prevent tampering with the firmware image before the image is loaded on a phone. Tampering with the image causes a phone to fail the authentication process and reject the new image.

Feature	Description
Image encryption	Encrypted binary files (with the extension.sebn) prevent tampering with the firmware image before the image is loaded on a phone. Tampering with the image causes a phone to fail the authentication process and reject the new image.
Customer site certificate installation	Each Cisco IP Phone requires a unique certificate for device authentication. Phones include a manufacturing installed certificate (MIC), but for additional security, you can specify certificate installation in Cisco Unified Communications Manager Administration using the Certificate Authority Proxy Function (CAPF). Alternatively, you can install a Locally Significant Certificate (LSC) from the Security Configuration menu on the phone.
Device authentication	Occurs between the Cisco Unified Communications Manager server and the phone when each entity accepts the certificate of the other entity. Determines whether a secure connection between the phone and a Cisco Unified Communications Manager should occur; and, if necessary, creates a secure signaling path between the entities by using TLS protocol. Cisco Unified Communications Manager does not register phones unless it can authenticate them.
File authentication	Validates digitally signed files that the phone downloads. The phone validates the signature to make sure that file tampering did not occur after file creation. Files that fail authentication are not written to Flash memory on the phone. The phone rejects such files without further processing.
File encryption	Encryption prevents sensitive information from being revealed while the file is in transit to the phone. In addition, the phone validates the signature to make sure that file tampering did not occur after file creation. Files that fail authentication are not written to Flash memory on the phone. The phone rejects such files without further processing.
Signaling authentication	Uses the TLS protocol to validate that no tampering to signaling packets has occurred during transmission.
Manufacturing installed certificate	Each Cisco IP Phone contains a unique manufacturing installed certificate (MIC), which is used for device authentication. The MIC provides permanent unique proof of identity for the phone and allows Cisco Unified Communications Manager to authenticate the phone.
Media encryption	Uses SRTP to ensure that media streams between supported devices prove secure and that only the intended device receives and reads the data. Includes creating a media primary key pair for the devices, delivering the keys to the devices, and securing the delivery of the keys while the keys are in transport.
CAPF (Certificate Authority Proxy Function)	Implements parts of the certificate generation procedure that are too processing-intensive for the phone, and interacts with the phone for key generation and certificate installation. The CAPF can be configured to request certificates from customer-specified certificate authorities on behalf of the phone, or it can be configured to generate certificates locally.
Security profile	Defines whether the phone is nonsecure, authenticated, encrypted, or protected. Other entries in this table describe security features.
Encrypted configuration files	Lets you ensure the privacy of phone configuration files.

Feature	Description
Optional web server disabling for a phone	For security purposes, you can prevent access to the web pages for a phone (which display a variety of operational statistics for the phone) and Self Care Portal.
Phone hardening	Additional security options, which you control from Cisco Unified Communications Manager Administration: • Disabling PC port • Disabling Gratuitous ARP (GARP) • Disabling PC Voice VLAN access • Disabling access to the Setting menus, or providing restricted access that allows access to the Preferences menu and saving volume changes only • Disabling access to web pages for a phone • Disabling Bluetooth Accessory Port • Restricting TLS ciphers
802.1X Authentication	The Cisco IP Phone can use 802.1X authentication to request and gain access to the network. See 802.1X Authentication, on page 106 for more information.
Secure SIP Failover for SRST	After you configure a Survivable Remote Site Telephony (SRST) reference for security and then reset the dependent devices in Cisco Unified Communications Manager Administration, the TFTP server adds the SRST certificate to the phone cnf.xml file and sends the file to the phone. A secure phone then uses a TLS connection to interact with the SRST-enabled router.
Signaling encryption	Ensures that all SIP signaling messages that are sent between the device and the Cisco Unified Communications Manager server are encrypted.
Trust List update alarm	When the Trust List updates on the phone, the Cisco Unified Communications Manager receives an alarm to indicate the success or failure of the update. See the following table for more information.

AES 256 Encryption	When connected to Cisco Unified Communications Manager Release 10.5(2) and later, the phones support AES 256 encryption support for TLS and SIP for signaling and media encryption. This enables phones to initiate and support TLS 1.2 connections using AES-256 based ciphers that conform to SHA-2 (Secure Hash Algorithm) standards and are Federal Information Processing Standards (FIPS) compliant. The ciphers include:
	• For TLS connections:
	  • TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384
	  • TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256
	• For sRTP:
	  • AEAD_AES_256_GCM
	  • AEAD_AES_128_GCM
	For more information, see the Cisco Unified Communications Manager documentation.

Elliptic Curve Digital Signature Algorithm (ECDSA) certificates	As part of Common Criteria (CC) certification, Cisco Unified Communications Manageradded ECDSA certificates in version 11.0. This affects all Voice Operating System (VOS) products from version CUCM 11.5 and later.

The following table contains the Trust List update alarm messages and meaning. For more information, see the Cisco Unified Communications Manager documentation

Table 25: Trust List Update Alarm Messages

Code and Message	Description
1 - TL_SUCCESS	Received new CTL and/or ITL
2 - CTL_INITIAL_SUCCESS	Received new CTL, no existing TL
3 - ITL_INITIAL_SUCCESS	Received new ITL, no existing TL
4 - TL_INITIAL_SUCCESS	Received new CTL and ITL, no existing TL
5 - TL_FAILED_OLD_CTL	Update to new CTL failed, but have previous TL
6 - TL_FAILED_NO_TL	Update to new TL failed, and have no old TL
7 - TL_FAILED	Generic failure
8 - TL_FAILED_OLD_ITL	Update to new ITL failed, but have previous TL
9 - TL_FAILED_OLD_TL	Update to new TL failed, but have previous TL

The Security Setup menu provides information about various security settings. The menu also provides access to the Trust List menu and indicates whether the CTL or ITL file is installed on the phone.

The following table describes the options in the Security Setup menu.

Table 26: Security Setup Menu

Option	Description	To Change
Security Mode	Displays the security mode that is set for the phone.	From Cisco Unified Communications Manager Administration, choose **Device** > **Phone**. The setting appears in the Protocol Specific Information portion of the Phone Configuration window.
LSC	Indicates whether a locally significant certificate that is used for security features is installed on the phone (Yes) or is not installed on the phone (No).	For information about how to manage the LSC for your phone, see the documentation for your particular Cisco Unified Communications Manager release.

Option	Description	To Change
Trust List	The Trust List provides submenus for the CTL, ITL, and Signed Configuration files. The CTL File submenu displays the contents of the CTL file. The ITL File submenu displays contents of the ITL file. The Trust List menu also displays the following information: • CTL Signature: the SHA1 hash of the CTL file • Unified CM/TFTP Server: the name of the Cisco Unified Communications Manager and TFTP Server that the phone uses. Displays a certificate icon if a certificate is installed for this server. • CAPF Server: the name of the CAPF server that the phone uses. Displays a certificate icon if a certificate is installed for this server. • SRST Router: the IP address of the trusted SRST router that the phone can use. Displays a certificate icon if a certificate is installed for this server.	For more information, see Set Up a Locally Significant Certificate, on page 87.
802.1X Authentication	Allows you to enable 802.1X authentication for this phone.	See 802.1X Authentication, on page 106.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Set Up a Locally Significant Certificate

This task applies to setting up a LSC with the authentication string method.

Before you begin

Make sure that the appropriate Cisco Unified Communications Manager and the Certificate Authority Proxy Function (CAPF) security configurations are complete:

- The CTL or ITL file has a CAPF certificate.
- In Cisco Unified Communications Operating System Administration, verify that the CAPF certificate is installed.
- The CAPF is running and configured.

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

Procedure

Step 1 Obtain the CAPF authentication code that was set when the CAPF was configured.

Step 2 From the phone, press **Applications** .

Step 3 Choose **Admin Settings** > **Security Setup**.

Note You can control access to the Settings menu by using the Settings Access field in the Cisco Unified Communications Manager Administration Phone Configuration window.

Step 4 Choose **LSC** and press **Select** or **Update**.

The phone prompts for an authentication string.

Step 5 Enter the authentication code and press **Submit**.

The phone begins to install, update, or remove the LSC, depending on how the CAPF is configured. During the procedure, a series of messages appears in the LSC option field in the Security Configuration menu, so you can monitor progress. When the procedure is complete, Installed or Not Installed displays on the phone.

The LSC install, update, or removal process can take a long time to complete.

When the phone installation procedure is successful, the `Installed` message displays. If the phone displays `Not Installed`, then the authorization string may be incorrect or the phone upgrade may not be enabled. If the CAPF operation deletes the LSC, the phone displays `Not Installed` to indicate that the operation succeeded. The CAPF server logs the error messages. See the CAPF server documentation to locate the logs and to understand the meaning of the error messages.

Enable FIPS Mode

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **Device** > **Phone** and locate the phone.

Step 2 Navigate to the Product Specific Configuration area.

Step 3 Set the **FIPS Mode** field to Enabled.

Step 4 Select **Apply Config**.

Step 5 Select **Save**.

Step 6 Restart the phone.

Phone Call Security

When security is implemented for a phone, you can identify secure phone calls by icons on the phone screen. You can also determine whether the connected phone is secure and protected if a security tone plays at the beginning of the call.

In a secure call, all call signaling and media streams are encrypted. A secure call offers a high level of security, providing integrity and privacy to the call. When a call in progress is encrypted, the call progress icon to the right of the call duration timer in the phone screen changes to the following icon: .

Note If the call is routed through non-IP call legs, for example, PSTN, the call may be nonsecure even though it is encrypted within the IP network and has a lock icon associated with it.

In a secure call, a security tone plays at the beginning of a call to indicate that the other connected phone is also receiving and transmitting secure audio. If your call connects to a nonsecure phone, the security tone does not play.

Note Secure calling is supported for connections between two phones only. Some features, such as conference calling and shared lines, are not available when secure calling is configured.

When a phone is configured as secure (encrypted and trusted) in Cisco Unified Communications Manager, it can be given a "protected" status. After that, if desired, the protected phone can be configured to play an indication tone at the beginning of a call:

- Protected Device: To change the status of a secure phone to protected, check the Protected Device check box in the Phone Configuration window in Cisco Unified Communications Manager Administration (**Device** > **Phone**).
- Play Secure Indication Tone: To enable the protected phone to play a secure or nonsecure indication tone, set the Play Secure Indication Tone setting to True. By default, Play Secure Indication Tone is set to False. You set this option in Cisco Unified Communications Manager Administration (**System** > **Service Parameters**). Select the server and then the Unified Communications Manager service. In the Service Parameter Configuration window, select the option in the Feature - Secure Tone area. The default is False.

Secure Conference Call Identification

You can initiate a secure conference call and monitor the security level of participants. A secure conference call is established by using this process:

1. A user initiates the conference from a secure phone.
2. Cisco Unified Communications Manager assigns a secure conference bridge to the call.
3. As participants are added, Cisco Unified Communications Manager verifies the security mode of each phone and maintains the secure level for the conference.
4. The phone displays the security level of the conference call. A secure conference displays the secure icon  to the right of **Conference** on the phone screen.

Note Secure calling is supported between two phones. For protected phones, some features, such as conference calling, shared lines, and Extension Mobility, are not available when secure calling is configured.

The following table provides information about changes to conference security levels depending on the initiator phone security level, the security levels of participants, and the availability of secure conference bridges.

Table 27: Security Restrictions with Conference Calls

Initiator Phone Security Level	Feature Used	Security Level of Participants	Results of Action
Nonsecure	Conference	Secure	Nonsecure conference bridge Nonsecure conference
Secure	Conference	At least one member is nonsecure.	Secure conference bridge Nonsecure conference
Secure	Conference	Secure	Secure conference bridge Secure encrypted level conference
Nonsecure	Meet Me	Minimum security level is encrypted.	Initiator receives message `Does not meet Security Level, call rejected`.
Secure	Meet Me	Minimum security level is nonsecure.	Secure conference bridge Conference accepts all calls.

Secure Phone Call Identification

A secure call is established when your phone, and the phone on the other end, is configured for secure calling. The other phone can be in the same Cisco IP network, or on a network outside the IP network. Secured calls can only be made between two phones. Conference calls should support secure call after secure conference bridge set up.

A secured call is established using this process:

1. A user initiates the call from a secured phone (secured security mode).
2. The phone displays the secure icon on the phone screen. This icon indicates that the phone is configured for secure calls, but this does not mean that the other connected phone is also secured.
3. The user hears a security tone if the call connects to another secured phone, indicating that both ends of the conversation are encrypted and secured. If the call connects to a nonsecure phone, the user does not hear the security tone.

Note Secure calling is supported between two phones. For protected phones, some features, such as conference calling, shared lines, and Extension Mobility, are not available when secure calling is configured.

Only protected phones play these secure or nonsecure indication tones. Nonprotected phones never play tones. If the overall call status changes during the call, the indication tone changes and the protected phone plays the appropriate tone.

A protected phone plays a tone or not under these circumstances:

- When the Play Secure Indication Tone option is enabled:
  - When end-to-end secure media is established and the call status is secure, the phone plays the secure indication tone (three long beeps with pauses).
  - When end-to-end nonsecure media is established and the call status is nonsecure, the phone plays the nonsecure indicationtone (six short beeps with brief pauses).

If the Play Secure Indication Tone option is disabled, no tone plays.

Provide Encryption for Barge

Cisco Unified Communications Manager checks the phone security status when conferences are established and changes the security indication for the conference or blocks the completion of the call to maintain integrity and security in the system.

A user cannot barge into an encrypted call if the phone that is used to barge is not configured for encryption. When barge fails in this case, a reorder (fast busy) tone plays on the phone that the barge was initiated.

If the initiator phone is configured for encryption, the barge initiator can barge into a nonsecure call from the encrypted phone. After the barge occurs, Cisco Unified Communications Manager classifies the call as nonsecure.

If the initiator phone is configured for encryption, the barge initiator can barge into an encrypted call, and the phone indicates that the call is encrypted.

WLAN Security

Because all WLAN devices that are within range can receive all other WLAN traffic, securing voice communications is critical in WLANs. To ensure that intruders do not manipulate nor intercept voice traffic, the Cisco SAFE Security architecture supports the Cisco IP Phone and Cisco Aironet APs. For more information about security in networks, see http://www.cisco.com/en/US/netsol/ns744/networking_solutions_program_home.html.

The Cisco Wireless IP telephony solution provides wireless network security that prevents unauthorized sign-ins and compromised communications by using the following authentication methods that the wireless Cisco IP Phone supports:

- Open Authentication: Any wireless device can request authentication in an open system. The AP that receives the request may grant authentication to any requestor or only to requestors that are found on a list of users. Communication between the wireless device and AP could be nonencrypted or devices can use Wired Equivalent Privacy (WEP) keys to provide security. Devices that use WEP only attempt to authenticate with an AP that is using WEP.
- Extensible Authentication Protocol-Flexible Authentication via Secure Tunneling (EAP-FAST) Authentication: This client server security architecture encrypts EAP transactions within a Transport Level Security (TLS) tunnel between the AP and the RADIUS server, such as the Cisco Access Control Server (ACS).

  The TLS tunnel uses Protected Access Credentials (PACs) for authentication between the client (phone) and the RADIUS server. The server sends an Authority ID (AID) to the client (phone), which in turn selects the appropriate PAC. The client (phone) returns a PAC-Opaque to the RADIUS server. The server decrypts the PAC with the primary key. Both endpoints now contain the PAC key and a TLS tunnel is created. EAP-FAST supports automatic PAC provisioning, but you must enable it on the RADIUS server.

Note In the Cisco ACS, by default, the PAC expires in one week. If the phone has an expired PAC, authentication with the RADIUS server takes longer while the phone gets a new PAC. To avoid PAC provisioning delays, set the PAC expiration period to 90 days or longer on the ACS or RADIUS server.

- Extensible Authentication Protocol-Transport Layer Security (EAP-TLS) Authentication: EAP-TLS requires a client certificate for authentication and network access. For wired EAP-TLS, the client certificate can be either the phone's MIC or an LSC. LSC is the recommended client authentication certificate for wired EAP-TLS.
- Protected Extensible Authentication Protocol (PEAP): Cisco proprietary password-based mutual authentication scheme between the client (phone) and a RADIUS server. Cisco IP Phone can use PEAP for authentication with the wireless network. Both PEAP-MSCHAPV2 and PEAP-GTC authentication methods are supported.

The following authentication schemes use the RADIUS server to manage authentication keys:

- WPA/WPA2: Uses RADIUS server information to generate unique keys for authentication. Because these keys are generated at the centralized RADIUS server, WPA/WPA2 provides more security than WPA preshared keys that are stored on the AP and phone.
- Fast Secure Roaming: Uses RADIUS server and a wireless domain server (WDS) information to manage and authenticate keys. The WDS creates a cache of security credentials for CCKM-enabled client devices for fast and secure reauthentication. The Cisco IP Phone 8800 Series supports 802.11r (FT). Both 11r (FT) and CCKM are supported to allow for fast secure roaming. But Cisco strongly recommends to utilize the 802.11r (FT) over air method.

With WPA/WPA2 and CCKM, encryption keys are not entered on the phone, but are automatically derived between the AP and phone. But the EAP username and password that are used for authentication must be entered on each phone.

To ensure that voice traffic is secure, the Cisco IP Phone supports WEP, TKIP, and Advanced Encryption Standards (AES) for encryption. When these mechanisms are used for encryption, both the signalling SIP packets and voice Real-Time Transport Protocol (RTP) packets are encrypted between the AP and the Cisco IP Phone.

WEP

With WEP use in the wireless network, authentication happens at the AP by using open or shared-key authentication. The WEP key that is setup on the phone must match the WEP key that is configured at the AP for successful connections. The Cisco IP Phone supports WEP keys that use 40-bit encryption or a 128-bit encryption and remain static on the phone and AP.

EAP and CCKM authentication can use WEP keys for encryption. The RADIUS server manages the WEP key and passes a unique key to the AP after authentication for encrypting all voice packets; consequently, these WEP keys can change with each authentication.

TKIP

WPA and CCKM use TKIP encryption that has several improvements over WEP. TKIP provides per-packet key ciphering and longer initialization vectors (IVs) that strengthen encryption. In addition, a message integrity check (MIC) ensures that encrypted packets are not being altered. TKIP removes the predictability of WEP that helps intruders decipher the WEP key.

AES

An encryption method used for WPA2 authentication. This national standard for encryption uses a symmetrical algorithm that has the same key for encryption and decryption. AES uses Cipher Blocking Chain (CBC) encryption of 128 bits in size, which supports key sizes of 128, 192 and 256 bits, as a minimum. The Cisco IP Phone supports a key size of 256 bits.

Note The Cisco IP Phone does not support Cisco Key Integrity Protocol (CKIP) with CMIC.

Authentication and encryption schemes are set up within the wireless LAN. VLANs are configured in the network and on the APs and specify different combinations of authentication and encryption. An SSID associates with a VLAN and the particular authentication and encryption scheme. In order for wireless client devices to authenticate successfully, you must configure the same SSIDs with their authentication and encryption schemes on the APs and on the Cisco IP Phone.

Some authentication schemes require specific types of encryption. With Open authentication, you can use static WEP for encryption for added security. But if you are using Shared Key authentication, you must set static WEP for encryption, and you must configure a WEP key on the phone.

Note

- When you use WPA pre-shared key or WPA2 pre-shared key, the pre-shared key must be statically set on the phone. These keys must match the keys that are on the AP.
- The Cisco IP Phone does not support auto EAP negotiation; to use EAP-FAST mode, you must specify it.

The following table provides a list of authentication and encryption schemes that are configured on the Cisco Aironet APs that the Cisco IP Phone supports. The table shows the network configuration option for the phone that corresponds to the AP configuration.

Table 28: Authentication and Encryption Schemes

Cisco IP Phone Configuration	**AP Configuration**			
Security Mode	Security	Key Management	Encryption	Fast Roaming
None	None	None	None	N/A
WEP	Static WEP	Static	WEP	N/A
PSK	PSK	WPA	TKIP	None
		WPA2	AES	FT
EAP-FAST	EAP-FAST	802.1x	WEP	CCKM
		WPA	TKIP	CCKM
		WPA2	AES	FT, CCKM

Cisco IP Phone Configuration	AP Configuration			
EAP-TLS	EAP-TLS	802.1x	WEP	CCKM
		WPA	TKIP	CCKM
		WPA2	AES	FT, CCKM
PEAP-MSCHAPV2	PEAP-MSCHAPV2	802.1x	WEP	CCKM
		WPA	TKIP	CCKM
		WPA2	AES	FT, CCKM
PEAP-GTC	PEAP-GTC	802.1x	WEP	CCKM
		WPA	TKIP	CCKM
		WPA2	AES	FT, CCKM

For more information about configuring authentication and encryption schemes on APs, see the *Cisco Aironet Configuration Guide* for your model and release under the following URL:

http://www.cisco.com/cisco/web/psa/configure.html?mode=prod&level0=278875243

Set Up Authentication Mode

To select the Authentication Mode for this profile, follow these steps:

Procedure

Step 1 Choose the network profile that you want to configure.

Step 2 Choose the authentication mode.

Note Depending on what you selected, you must configure additional options in Wireless Security or Wireless Encryption. See WLAN Security, on page 91 for more information.

Step 3 Click **Save** to make the change.

Wireless Security Credentials

When your network uses EAP-FAST and PEAP for user authentication, you must configure both the username and password if required on the Remote Authentication Dial-In User Service (RADIUS) and the phone.

Note If you use domains within your network, you must enter the username with the domain name, in the format: *domain\username*.

The following actions could result in the existing Wi-Fi password being cleared:

- Entering an invalid user id or password
- Installing an invalid or expired Root CA when the EAP type is set to PEAP-MSCHAPV2 or PEAP-GTC
- Disabling the EAP type on the RADIUS server used by the phone before changing a phone to the new EAP type

To change EAP types, do the following in the stated order:

- Enable the new EAP types on the RADIUS.
- Change the EAP type on a phone to the new EAP type.

Keep the current EAP type configured on the phone until the new EAP type is enabled on the RADIUS server. Once the new EP type is enabled on the RADIUS server, then you can change the phone's EAP type. Once all phones have been changed to the new EAP type, you can disable the previous EAP type if you want.

Set Up Username and Password

To enter or change the username or password for the network profile, you must use the same username and the same password string that are configured in the RADIUS server. The maximum length of the username or password entry is 64 characters.

To set up the username and password in Wireless Security Credentials, follow these steps:

Procedure

Step 1 Choose the network profile.

Step 2 In the Username field, enter the network username for this profile.

Step 3 In the Password field, enter the network password string for this profile.

Step 4 Click **Save** to make the change.

Pre-Shared Key Setup

Use the following sections to guide you as you set up pre-shared keys.

Pre-Shared Key Formats

The Cisco IP Phone supports ASCII and hexadecimal formats. You must use one of these formats when setting up a WPA Pre-shared key:

Hexadecimal

For hexadecimal keys, you enter 64 hex digits (0-9 and A-F); for example, AB123456789CD01234567890EFAB123456789CD01234567890EF3456789C

ASCII

For ASCII keys, you enter a character string that uses 0-9, A-Z (upper and lower case), including symbols and is from 8 to 63 characters in length; for example, GREG12356789ZXYW

Set Up PSK

To set up a PSK in the Wireless Credentials area, follow these steps:

Procedure

Step 1 Choose the network profile that enables the WPA Pre-shared key or WPA2 Pre-shared key.

Step 2 In the Key Type area, enter the appropriate key.

Step 3 Enter an ASCII string or hexadecimal digits in the Passphrase/Pre-shared key field.

Step 4 Click **Save** to make the change.

Wireless Encryption

If your wireless network uses WEP encryption, and you set the Authentication Mode as Open + WEP, you must enter an ASCII or hexadecimal WEP Key.

The WEP Keys for the phone must match the WEP Keys assigned to the access point. Cisco IP Phone and Cisco Aironet Access Points support both 40-bit and 128-bit encryption keys.

WEP Key Formats

You must use one of these formats when setting up a WEP key:

Hexadecimal

For hexadecimal keys, you use one of the following key sizes:

40-bit

You enter a 10-digit encryption key string that uses the hex digits (0-9 and A-F); for example, ABCD123456.

128-bit

You enter a 26-digit encryption key string that uses the hex digits (0-9 and A-F); for example, AB123456789CD01234567890EF.

ASCII

For ASCII keys, you enter a character string that uses 0-9, A-Z (upper and lower case), and all symbols, with one of the following key sizes:

40-bit

You enter a 5-character string; for example, GREG5.

128-bit

You enter a 13-character string; for example, GREGSSECRET13.

Set Up WEP Keys

To set up WEP keys, follow these steps.

Procedure

Step 1 Choose the network profile that uses Open+WEP or Shared+WEP.

Step 2 In the Key Type area, enter the appropriate key.

Step 3 In the Key Size area, choose one of these character string lengths:

- **40**
- **128**

Step 4 In the Encryption Key field, enter the appropriate key string based on the selected Key Type and Key Size. See WEP Key Formats, on page 96.

Step 5 Click **Save** to make the change.

Export CA certificate from ACS using Microsoft Certificate Services

Export the root CA certificate from the ACS server. For additional information, see the CA or RADIUS documentation.

Manufacturing Installed Certificate

Cisco has included a Manufacturing Installed Certificate (MIC) in the phone at the factory.

During EAP-TLS authentication, the ACS server needs to verify the trust of the phone and the phone needs to verify the trust of the ACS server.

To verify the MIC, the Manufacturing Root Certificate and Manufacturing Certificate Authority (CA) Certificate must be exported from a Cisco IP Phone and installed on the Cisco ACS server. These two certificates are part of the trusted certificate chain used to verify the MIC by the Cisco ACS server.

To verify the Cisco ACS certificate, a trusted subordinate certificate (if any) and root certificate (created from a CA) on the Cisco ACS server must be exported and installed on the phone. These certificates are part of the trusted certificate chain used to verify the trust of the certificate from the ACS server.

User-Installed Certificate

To use a user-installed certificate, a Certificate Signing Request (CSR) is generated, and sent to the CA for approval. A user certificate can also be generated by the CA without a CSR.

During EAP-TLS authentication, the ACS server verifies the trust of the phone and the phone verifies the trust of the ACS server.

To verify the authenticity of the user-installed certificate, you must install a trusted subordinate certificate (if any) and root certificate from the CA that approved the user certificate on the Cisco ACS server. These certificates are part of the trusted certificate chain used to verify the trust of the user installed certificate.

To verify the Cisco ACS certificate, you export a trusted subordinate certificate (if any) and root certificate (created from a CA) on the Cisco ACS server and the exported certificates are installed on the phone. These certificates are part of the trusted certificate chain used to verify the trust of the certificate from the ACS server.

Install EAP-TLS Authentication Certificates

To install authentication certificates for EAP-TLS, perform the following steps.

Procedure

Step 1 From the phone web page, set the Cisco Unified Communications Manager date and time on the phone.

Step 2 If using the Manufacturing Installed Certificate (MIC):

a) From the phone web page, export the CA root certificate and manufacturing CA certificate.
b) From Internet Explorer, install certificates on the CiscoACS server and edit the trust list.
c) Import the root CA to the phone.

For more information, see:

- Export and Install Certificates on ACS, on page 99
- Export CA certificate from ISE using Microsoft Certificate Services, on page 99

Step 3 Using the ACS configuration tool, set up the user account.

For more information, see:

- Set Up ACS User Account and Install Certificate, on page 101
- *User Guide for Cisco Secure ACS for Windows*(http://www.cisco.com/c/en/us/support/security/secure-access-control-system/products-user-guide-list.html)

Set Date and Time

EAP-TLS uses certificate-based authentication that requires the internal clock on the Cisco IP Phone to be set correctly. The date and time on the phone might change when it is registered to CiscoUnified Communications Manager.

Note If a new server authentication certificate is being requested and the local time is behind the Greenwich Mean Time (GMT), the authentication certificate validation might fail. Cisco recommends that you set the local date and time ahead of the GMT.

To set the phone to the correct local date and time, follow these steps.

Procedure

Step 1 Select **Date & Time** from the left navigation pane.

Step 2 If the setting in the Current Phone Date & Time field is different from the Local Date & Time field, click **Set Phone to Local Date & Time**.

Step 3 Click **Phone Restart**, and then click **OK**.

Export and Install Certificates on ACS

To use the MIC, export the Manufacturing Root Certificate and Manufacturing CA Certificate and install it on the Cisco ACS server.

To export the manufacturing root certificate and manufacturing CA certificate to the ACS server, follow these steps.

Procedure

Step 1 From the phone web page, choose **Certificates**.

Step 2 Click **Export** next to the Manufacturing Root Certificate.

Step 3 Save the certificate and copy it to the ACS server.

Step 4 Repeat Steps 1 and 2 for the Manufacturing CA certificate.

Step 5 From the ACS Server System Configuration page, enter the file path for each certificate and install the certificates.

Note For more information about using the ACS configuration tool, see the ACS online help or the *User Guide for Cisco Secure ACS for Windows*(http://www.cisco.com/c/en/us/support/security/secure-access-control-system/products-user-guide-list.html).

Step 6 Use the Edit the Certificate Trust List (CTL) page to add the certificates to be trusted by ACS.

ACS Certificate Export Methods

Depending on the type of certificate you export from the ACS, use one of the following methods:

- To export the CA certificate from the ACS server that signed the user-installed certificate or ACS certificate, see Export CA certificate from ISE using Microsoft Certificate Services, on page 99.
- To export the CA certificate from the ACS server that uses a self-signed certificate, see Export CA Certificate from ACS Using Internet Explorer, on page 100.

Export CA certificate from ISE using Microsoft Certificate Services

Use this method to export the CA certificate from the ISE server that signed the user-installed certificate or ISE certificate.

To export the CA certificate using the Microsoft Certificate Services web page, follow these steps.

Procedure

Step 1 From the Microsoft Certificate Services web page, select **Download a CA certificate, certificate chain or CRL**.

Step 2 At the next page, highlight the current CA certificate in the text box, choose DER under Encoding Method, then click **Download CA certificate**.

Step 3 Save the CA certificate.

Export CA Certificate from ACS Using Internet Explorer

Use this method to export the CA certificate from the ACS server that uses a self-signed certificate.

To export certificates from the ACS server using Internet Explorer, follow these steps.

Procedure

Step 1 From Internet Explorer, choose **Tools** > **Internet Options**, then click the Content tab.

Step 2 Under Certificates, click **Certificates**, then click the Trusted Root Certification Authorities tab.

Step 3 Highlight the root certificate and click **Export**. The Certificate Export Wizard appears.

Step 4 Click **Next**.

Step 5 At the next window, select **DER encoded binary X.509 (.CER)**, and click **Next**.

Step 6 Specify a name for the certificate and click **Next**.

Step 7 Save the CA certificate to be installed on the phone.

Request and Import User-Installed Certificate

To request and install the certificate on the phone, follow these steps.

Procedure

Step 1 From the phone web page, choose the network profile using EAP-TLS, and select User Installed in the EAP-TLS Certificate field.

Step 2 Click **Certificates**.

On the User Certificate Installation page, the Common Name field should match the user name in the ACS server.

Note You can edit the Common Name field if you wish. Make sure that it matches the username in the ACS server. See Set Up ACS User Account and Install Certificate, on page 101.

Step 3 Enter the information to be displayed on the certificate, and click **Submit** to generate the Certificate Signing Request (CSR).

Install Authentication Server Root Certificate

To install the Authentication Server Root Certificate on the phone, follow these steps.

Procedure

Step 1 Export the Authentication Server Root Certificate from the ACS. See ACS Certificate Export Methods, on page 99.

Step 2 Go to the phone web page and choose **Certificates**.

Step 3 Click **Import** next to the Authentication Server Root certificate.

Step 4 Restart the phone.

Set Up ACS User Account and Install Certificate

To set up the user account name and install the MIC root certificate for the phone on the ACS, follow these steps.

Note For more information about using the ACS configuration tool, see the ACS online help or the *User Guide for Cisco Secure ACS for Windows* .

Procedure

Step 1 From the ACS configuration tool User Setup page, create a phone user account name if it is not already set up.

Typically, the user name includes the phone MAC address at the end. No password is necessary for EAP-TLS.

Note Make sure the user name matches the Common Name field in the User Certificate Installation page. See Request and Import User-Installed Certificate, on page 100.

Step 2 On the System Configuration page, in the EAP-TLS section, enable these fields:

- **Allow EAP-TLS**
- **Certificate CN comparison**

Step 3 On the ACS Certification Authority Setup page, add the Manufacturing Root Certificate and Manufacturing CA Certificate to the ACS server.

Step 4 Enable both the Manufacturing Root Certificate and Manufacturing CA Certificate in the ACS Certificate Trust List.

PEAP Setup

Protected Extensible Authentication Protocol (PEAP) uses server-side public key certificates to authenticate clients by creating an encrypted SSL/TLS tunnel between the client and the authentication server.

Cisco IP Phone 8865 only supports one server certificate which can be installed either via SCEP or the manual install method but not both. The phone does not support the TFTP method of certificate installation.

Note The authentication server validation can be enabled by importing the authentication server certificate.

Before You Begin

Before you configure PEAP authentication for the phone, make sure these CiscoSecure ACS requirements are met:

- The ACS root certificate must be installed.
- A certificate can also be installed to enable Server Validation for PEAP. But if a server certificate is installed then server validation is enabled.
- The Allow EAP-MSCHAPv2 setting must be enabled.
- User account and password must be configured.
- For password authentication, you can use the local ACS database or an external one (such as Windows or LDAP).

Enable PEAP Authentication

Procedure

Step 1 From the phone configuration web page, choose PEAP as the authentication mode.

Step 2 Enter a user name and password.

Wireless LAN Security

Cisco phones that support Wi-Fi have more security requirements and require extra configuration. These extra steps include installing certificates and setting up security on the phones and on the Cisco Unified Communications Manager.

For additional information, see *Security Guide for Cisco Unified Communications Manager*.

Cisco IP Phone Administration Page

Cisco phones that support Wi-Fi have special web pages that are different from the pages for other phones. You use these special web pages for phone security configuration when Simple Certificate Enrollment Protocol (SCEP) is not available. Use these pages to manually install security certificates on a phone, to download a security certificate, or to manually configure the phone date and time.

These web pages also show the same information that you see on other phone web pages, including device information, network setup, logs, and statistical information.

Related Topics

Cisco IP Phone Web Page, on page 222

Configure the Administration Page for Phone

The administration web page is enabled when the phone ships from the factory and the password is set to Cisco. But if a phone registers with Cisco Unified Communications Manager, the administration web page must be enabled and a new password configured.

Enable this web page and set the sign-in credentials before you use the web page for the first time after the phone has registered.

Once enabled, the administration web page is accessible at HTTPS port 8443 (**`https://x.x.x.x:8443`**, where x.x.x.x is a phone IP address).

Before you begin

Decide on a password before you enable the administration web page. The password can be any combination of letters or numbers, but it must be between 8 and 127 characters in length.

Your username is permanently set to admin.

Procedure

Step 1 From the Cisco Unified Communications Manager Administration, select **Device** > **Phone**.

Step 2 Locate your phone.

Step 3 In the **Product Specific Configuration Layout** section, set **Web Admin** to **Enabled**.

Step 4 In the **Admin Password** field, enter a password.

Step 5 Select **Save** and click **OK**.

Step 6 Select **Apply Config** and click **OK**.

Step 7 Restart the phone.

Access the Phone Administration Web Page

When you want to access the administration web pages, you need to specify the administration port.

Procedure

Step 1 Obtain the IP address of the phone:

- In Cisco Unified Communications Manager Administration, select **Device** > **Phone**, and locate the phone. Phones that register with Cisco Unified Communications Manager display the IP address on the **Find and List Phones** window and at the top of the **Phone Configuration** window.
- On the phone, press **Applications** , choose **Phone Information**, and then scroll to the IPv4 address field.

Step 2 Open a web browser and enter the following URL, where *IP_address* is the IP address of the Cisco IP Phone:

```
https://<IP_address>:8443
```

Step 3 Enter the password in the Password field.

Step 4 Click **Submit**.

Install a User Certificate from the Phone Administration Web Page

You can manually install a user certificate on the phone if Simple Certificate Enrollment Protocol (SCEP) is not available.

The preinstalled Manufacturing Installed Certificate (MIC) can be used as the User Certificate for EAP-TLS.

After the User Certificate installs, you need to add it to the RADIUS server trust list.

Before you begin

Before you can install a User Certificate for a phone, you must have:

- A User Certificate saved on your PC. The certificate must be in PKCS #12 format.
- The certificate's extract password.

Procedure

Step 1 From the phone administration web page, select **Certificates**.
Step 2 Locate the User install field and click **Install**.
Step 3 Browse to the certificate on your PC.
Step 4 In the **Extract password** field, enter the certificate extract password.
Step 5 Click **Upload**.
Step 6 Restart the phone after the upload is complete.

Install an Authentication Server Certificate from the Phone Administration Web Page

You can manually install an Authentication Server certificate on the phone if Simple Certificate Enrollment Protocol (SCEP) is not available.

The root CA certificate that issued the RADIUS server certificate must be installed for EAP-TLS.

Before you begin

Before you can install a certificate on a phone, you must have an Authentication Server Certificate saved on your PC. The certificate must be encoded in PEM (Base-64) or DER.

Procedure

Step 1 From the phone administration web page, select **Certificates**.
Step 2 Locate the **Authentication server CA (Admin webpage)** field and click **Install**.
Step 3 Browse to the certificate on your PC.
Step 4 Click **Upload**.
Step 5 Restart the phone after the upload is complete.

If you are installing more than one certificate, install all of the certificates before restarting the phone.

Manually Remove a Security Certificate from the Phone Administration Web Page

You can manually remove a security certificate from a phone if Simple Certificate Enrollment Protocol (SCEP) is not available.

Procedure

Step 1 From the phone administration web page, select **Certificates**.

Step 2 Locate the certificate on the **Certificates** page.

Step 3 Click **Delete**.

Step 4 Restart the phone after the deletion process completes.

Manually Set the Phone Date and Time

With certificate-based authentication, the phone must display the correct date and time. An authentication server checks the phone date and time against the certificate expiry date. If the phone and the server dates and times don't match, the phone stops working.

Use this procedure to manually set the date and time on the phone if the phone is not receiving the correct information from your network.

Procedure

Step 1 From the phone administration web page, scroll to **Date & Time**.

Step 2 Perform one of the following options:

- Click **Set Phone to Local Date & Time** to synch the phone to a local server.
- In the **Specify Date & Time** fields, select the month, day, year, hour, minute, and second using the menus and click **Set Phone to Specific Date & Time**.

SCEP Setup

Simple Certificate Enrollment Protocol (SCEP) is the standard for automatically provisioning and renewing certificates. It avoids manual installation of certificates on your phones.

Configure the SCEP Product Specific Configuration Parameters

You must configure the following SCEP parameters on your phone web page

- RA IP address
- SHA-1 or SHA-256 fingerprint of the root CA certificate for the SCEP server

The Cisco IOS Registration Authority (RA) serves as a proxy to the SCEP server. The SCEP client on the phone use the parameters that are downloaded from Cisco Unified Communication Manager. After you configure the parameters, the phone sends a `SCEP getcs` request to the RA and the root CA certificate is validated using the defined fingerprint.

Procedure

Step 1 From the Cisco Unified Communications Manager Administration, select **Device** > **Phone**.

Step 2 Locate the phone.

Step 3 Scroll to the **Product Specific Configuration Layout** area.

Step 4 Check the **WLAN SCEP Server** check box to activate the SCEP parameter.

Step 5 Check the **WLAN Root CA Fingerprint (SHA256 or SHA1)** check box to activate the SCEP QED parameter.

Simple Certificate Enrollment Protocol Server Support

If you are using a Simple Certificate Enrollment Protocol (SCEP) server, the server can automatically maintain your user and server certificates. On the SCEP server, configure the SCEP Registration Agent (RA) to:

- Act as a PKI trust point
- Act as a PKI RA
- Perform device authentication using a RADIUS server

For more information, see your SCEP server documentation.

802.1X Authentication

The Cisco IP Phones support 802.1X Authentication.

Cisco IP Phones and Cisco Catalyst switches traditionally use Cisco Discovery Protocol (CDP) to identify each other and determine parameters such as VLAN allocation and inline power requirements. CDP does not identify locally attached workstations. Cisco IP Phones provide an EAPOL pass-through mechanism. This mechanism allows a workstation attached to the Cisco IP Phone to pass EAPOL messages to the 802.1X authenticator at the LAN switch. The pass-through mechanism ensures that the IP phone does not act as the LAN switch to authenticate a data endpoint before accessing the network.

Cisco IP Phones also provide a proxy EAPOL Logoff mechanism. In the event that the locally attached PC disconnects from the IP phone, the LAN switch does not see the physical link fail, because the link between the LAN switch and the IP phone is maintained. To avoid compromising network integrity, the IP phone sends an EAPOL-Logoff message to the switch on behalf of the downstream PC, which triggers the LAN switch to clear the authentication entry for the downstream PC.

Support for 802.1X authentication requires several components:

- Cisco IP Phone: The phone initiates the request to access the network. Cisco IP Phones contain an 802.1X supplicant. This supplicant allows network administrators to control the connectivity of IP phones to the LAN switch ports. The current release of the phone 802.1X supplicant uses the EAP-FAST and EAP-TLS options for network authentication.
- Cisco Secure Access Control Server (ACS) (or other third-party authentication server): The authentication server and the phone must both be configured with a shared secret that authenticates the phone.
- Cisco Catalyst Switch (or other third-party switch): The switch must support 802.1X, so it can act as the authenticator and pass the messages between the phone and the authentication server. After the exchange completes, the switch grants or denies the phone access to the network.

You must perform the following actions to configure 802.1X.

- Configure the other components before you enable 802.1X Authentication on the phone.

- Configure PC Port: The 802.1X standard does not consider VLANs and thus recommends that only a single device should be authenticated to a specific switch port. However, some switches (including Cisco Catalyst switches) support multidomain authentication. The switch configuration determines whether you can connect a PC to the PC port of the phone.
  - Enabled: If you are using a switch that supports multidomain authentication, you can enable the PC port and connect a PC to it. In this case, Cisco IP Phones support proxy EAPOL-Logoff to monitor the authentication exchanges between the switch and the attached PC. For more information about IEEE 802.1X support on the Cisco Catalyst switches, see the Cisco Catalyst switch configuration guides at:

    http://www.cisco.com/en/US/products/hw/switches/ps708/tsd_products_support_series_home.html
  - Disabled: If the switch does not support multiple 802.1X-compliant devices on the same port, you should disable the PC Port when 802.1X authentication is enabled. If you do not disable this port and subsequently attempt to attach a PC to it, the switch denies network access to both the phone and the PC.
- Configure Voice VLAN: Because the 802.1X standard does not account for VLANs, you should configure this setting based on the switch support.
  - Enabled: If you are using a switch that supports multidomain authentication, you can continue to use the voice VLAN.
  - Disabled: If the switch does not support multidomain authentication, disable the Voice VLAN and consider assigning the port to the native VLAN.

Access 802.1X Authentication

You can access the 802.1X authentication settings by following these steps:

Procedure

Step 1 Press **Applications** .

Step 2 Choose **Admin settings** > **Security setup** > **802.1X Authentication**.

Step 3 Configure the Options as described in 802.1X Authentication Options, on page 107.

Step 4 To exit this menu, press **Exit**.

802.1X Authentication Options

The following table describes the 802.1X authentication options.

Table 29: 802.1X Authentication Settings

Option	Description	To change
Device Authentication	Determines whether 802.1X authentication is enabled: • Enabled: Phone uses 802.1X authentication to request network access. • Disabled: Default setting. The phone uses CDP to acquire VLAN and network access.	See Set Device Authentication Field 108.
Transaction Status	State: Displays the state of 802.1x authentication: • Disconnected: Indicates that 802.1x authentication is not configured on the phone. • Authenticated: Indicates that the phone is authenticated. • Held: Indicates that the authentication process is in progress. Protocol: Displays the EAP method that is used for 802.1x authentication (can be EAP-FAST or EAP-TLS).	Display only. Cannot configure.

Set Device Authentication Field

Procedure

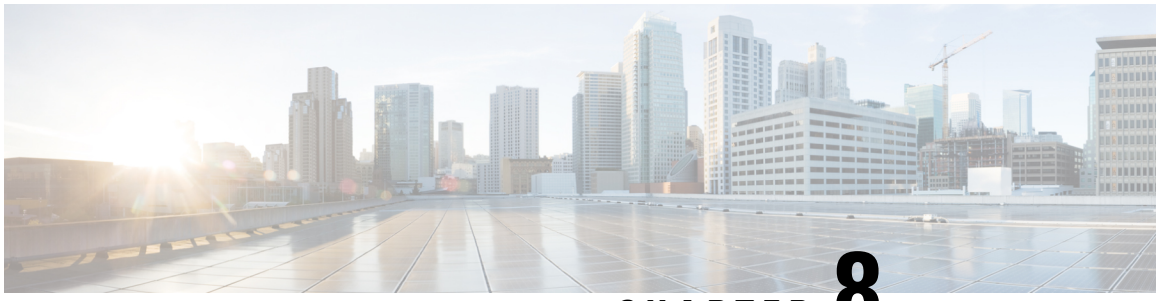
Step 1 Press **Applications** .

Step 2 Choose **Admin settings** > **Security setup** > **802.1X Authentication**

Step 3 Set the Device Authentication option:

- **Yes**
- **No**

Step 4 Press **Apply**.



CHAPTER 8

Cisco IP Phone Customization

- Custom Phone Rings, on page 109
- Custom Background Images, on page 109
- Set Up Wideband Codec, on page 111
- Set Up Idle Display, on page 111
- Customize the Dial Tone, on page 112

Custom Phone Rings

The phone ships with three ring tones that are implemented in hardware: Sunshine, Chirp, Chirp1.

Cisco Unified Communications Manager also provides a default set of additional phone ring sounds that are implemented in software as pulse code modulation (PCM) files. The PCM files, along with an XML file (named Ringlist-wb.xml) that describes the ring list options that are available at your site, exist in the TFTP directory on each Cisco Unified Communications Manager server.

Attention All file names are case sensitive. If you use Ringlist-wb.xml for the file name, the phone will not apply your changes.

For more information, see the "Custom Phone Rings and Backgrounds" chapter, Feature Configuration Guide for Cisco Unified Communications Manager for Cisco Unified Communications Manager release 12.0(1) or later.

Custom Background Images

You can customize a Cisco IP phone with a background image or wallpaper. Customized wallpapers are a popular way to display corporate logos or images and many organizations use them to make their phones stand out.

As of Firmware Release 12.7(1), you can customize your wallpaper on both your phones and your key expansion modules. But you need one image for the phone and one image for the expansion module.

The phone analyzes the wallpaper colors, and changes the font colors and icons so you can read them. If your wallpaper is dark, the phone changes the fonts and icons to white. If your wallpaper is light, the phone displays the fonts and icons as black.

It is best to choose a simple image such as a solid color or pattern for your background. Avoid high contrast images.

You add customized wallpaper in one of two ways:

- Using the List file
- Using a Common Phone Profile

If you want the user to be able to select your image from various wallpapers available on the phone, then modify the List file. But if you want to push the image to the phone, then create or modify an existing Common Phone Profile.

Regardless of your approach, note the following:

- Your images must be in PNG format and the full sized image must be within the following dimensions:
  - Thumbnail images—139 pixels (width) by 109 pixels (height)
  - Cisco IP Phone 8800 Series—800 pixels by 480 pixels
  - Cisco IP Phone 8851 and 8861 Key Expansion Module with a dual LCD screen—320 by 480 pixels
  - Cisco IP Phone 8865 Key Expansion Module with a dual LCD screen—320 by 480 pixels
  - Cisco IP Phone 8800 Key Expansion Module with a single LCD screen—272 by 480 pixels
- Upload the images, the thumbnails, and List file to your TFTP server. The directory is:
  - Cisco IP Phone 8800 Series—Desktops/800x480x24
  - Cisco IP Phone 8851 and 8861 Key Expansion Module with a dual LCD screen—Desktops/320x480x24
  - Cisco IP Phone 8865 Key Expansion Module with a dual LCD screen—Desktops/320x480x24
  - Cisco IP Phone 8800 Key Expansion Module with a single LCD screen—Desktops/272x480x24

  After the upload is done, you restart the TFTP server.
- If you don't want the user selecting their own wallpaper, then disable **Enable End User Access to Phone Background Image Setting**. Save and apply the phone profile. Restart the phones so your changes take effect.



Note You can apply the phone background images in bulk with the **Common Phone Profile**. But bulk configuration requires you to disable **Enable End User Access to Phone Background Image Setting**. For more information on bulk configuration of background images, refer to the “Configure the Common Phone Profile” chapter of Customized Wallpapers Best Practices Cisco IP Phone 8800 Series.)

For more information on customizing wallpaper, refer to the following documentation:

- Customized Wallpapers Best Practices Cisco IP Phone 8800 Series).

- "Custom Phone Rings and Backgrounds" chapter, Feature Configuration Guide for Cisco Unified Communications Manager for Cisco Unified Communications Manager release 12.0(1) or later.
- “Settings” chapter in the *Cisco IP Phone 8800 Series User Guide*.

Set Up Wideband Codec

By default, the G.722 codec is enabled for the Cisco IP Phone. If Cisco Unified Communications Manager is configured to use G.722 and if the far endpoint supports G.722, the call connects using the G.722 codec in place of G.711.

This situation occurs regardless of whether the user has enabled a wideband headset or wideband handset, but if either the headset or handset is enabled, the user may notice greater audio sensitivity during the call. Greater sensitivity means improved audio clarity but also means that the far endpoint can hear more background noise: noise such as rustling papers or nearby conversations. Even without a wideband headset or handset, some users may prefer the additional sensitivity of G.722 distracting. Other users may prefer the additional sensitivity of G.722.

The Advertise G.722 and iSAC Codec service parameter affects whether wideband support exists for all devices that register with this Cisco Unified Communications Manager server or for a specific phone, depending on the Cisco Unified Communications Manager Administration window where the parameter is configured.

Procedure

Step 1 To configure wideband support for all devices:

a) From Cisco Unified Communications Manager Administration, choose **System** > **Enterprise Parameters**
b) Set the Advertise G.722 and iSAC Codec field

The default value of this enterprise parameter is **True**, which means that all Cisco IP Phone Models that register to this Cisco Unified Communications Manager advertise G.722 to Cisco Unified Communications Manager. If each endpoint in the attempted call supports G.722 in the capabilities set, Cisco Unified Communications Manager chooses that codec for the call whenever possible.

Step 2 To configure wideband support for a specific device:

a) From Cisco Unified Communications Manager Administration, choose **Device** > **Phone**.
b) Set the Advertise G.722 and iSAC Codec parameter in the Product Specific Configuration area.

The default value of this product-specific parameter is to use the value that the enterprise parameter specifies. If you want to override this on a per-phone basis, choose **Enabled** or **Disabled**

Set Up Idle Display

You can specify an idle display (text only; text file size should not exceed 1M bytes) that appears on the phone screen. The idle display is an XML service that the phone invokes when the phone is idle (not in use) for a designated period and no feature menu is open.

For detailed instructions about creating and displaying the idle display, see *Creating Idle URL Graphics on Cisco IP Phone* at this URL:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a00801c0764.shtml

In addition, see the documentation for your particular Cisco Unified Communications Manager release for the following information:

- Specifying the URL of the idle display XML service:
  - For a single phone: Idle field in the Phone Configuration window in Cisco Unified Communications Manager Administration.
  - For multiple phones simultaneously: URL Idle field in the Enterprise Parameters Configuration window, or the Idle field in the Bulk Administration Tool (BAT)
- Specifying the length of time that the phone is not used before the idle display XML service is invoked:
  - For a single phone: Idle Timer field in the Phone configuration window in Cisco Unified Communications Manager Administration.
  - For multiple phones simultaneously: URL Idle Time field in the Enterprise Parameters Configuration window, or the Idle Timer field in the Bulk Administration Tool (BAT)

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **Device** > **Phone**

Step 2 In the Idle field, enter the URL to the idle display XML Service.

Step 3 In the Idle Timer field, enter the time that the idle phone waits before displaying the idle display XML service.

Step 4 Select **Save**.

Customize the Dial Tone

You can set up your phones so that users hear different dial tones for internal and external calls. Depending upon your needs, you can choose from three dial tone options:

- Default: A different dial tone for inside and outside calls.
- Inside: The inside dial tone is used for all calls.
- Outside: The outside dial tone is used for all calls.

Always Use Dial Tone is a required field on Cisco Unified Communications Manager.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **System** > **Service Parameters**.

Step 2 Select the appropriate Server.

Step 3 Select **Cisco CallManager** as the Service.

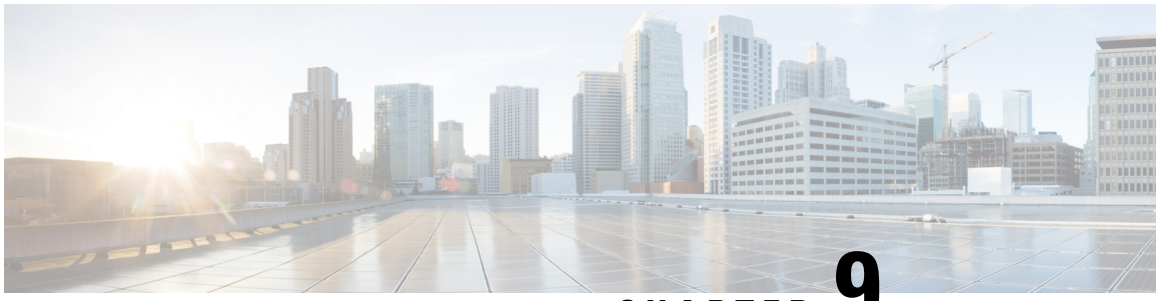
Step 4 Scroll to the Clusterwide Parameters pane.

Step 5 Set **Always Use Dial Tone** to one of the following:

- Outside
- Inside
- Default

Step 6 Select **Save**.

Step 7 Restart your phones.



CHAPTER 9

Phone Features and Setup

- Phone Features and Setup Overview, on page 115
- Cisco IP Phone User Support, on page 115
- Telephone Features, on page 116
- Feature Buttons and Softkeys, on page 133
- Phone Feature Configuration, on page 135
- Set Up Softkey Template, on page 183
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- Headset Management on Older Versions of Cisco Unified Communications Manager, on page 195

Phone Features and Setup Overview

After you install Cisco IP Phones in your network, configure their network settings, and add them to Cisco Unified Communications Manager, you must use the Cisco Unified Communications Manager Administration application to configure telephony features, optionally modify phone templates, set up services, and assign users.

You can modify additional settings for the Cisco IP Phone from Cisco Unified Communications Manager Administration. Use this web-based application to set up phone registration criteria and calling search spaces, to configure corporate directories and services, and to modify phone button templates, among other tasks.

When adding features to the phone line keys, you are limited by the number of line keys available. You cannot add more features than the number of line keys on your phone.

Cisco IP Phone User Support

If you are a system administrator, you are likely the primary source of information for Cisco IP Phone users in your network or company. It is important to provide current and thorough information to end users.

To successfully use some of the features on the Cisco IP Phone (including Services and voice message system options), users must receive information from you or from your network team or must be able to contact you

for assistance. Make sure to provide users with the names of people to contact for assistance and with instructions for contacting those people.

We recommend that you create a web page on your internal support site that provides end users with important information about their Cisco IP Phones.

Consider including the following types of information on this site:

- User guides for all Cisco IP Phone models that you support
- Information on how to access the Cisco Unified Communications Self Care Portal
- List of features supported
- User guide or quick reference for your voicemail system

Telephone Features

After you add Cisco IP Phones to Cisco Unified Communications Manager, you can add functionality to the phones. The following table includes a list of supported telephony features, many of which you can configure by using Cisco Unified Communications Manager Administration.

For information about using most of these features on the phone, see *Cisco IP Phone 8800 Series User Guide*. See Feature Buttons and Softkeys, on page 133 for a list of features that can be configured as programmable buttons and dedicated softkeys and feature buttons.

Note Cisco Unified Communications Manager Administration also provides several service parameters that you can use to configure various telephony functions. For more information on accessing and configuring service parameters, see the documentation for your particular Cisco Unified Communications Manager release.

For more information on the functions of a service, select the name of the parameter or the **question mark (?) help** button in the Product Specific Configuration window.

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

Feature	Description and more information
Abbreviated Dialing	Allows users to speed dial a phone number by entering an assigned index code (1-199) on the phone keypad. **Note** You can use Abbreviated Dialing while on-hook or off-hook. Users assign index codes from the Self Care Portal.
Actionable Incoming Call Alert	Provides different options to control the incoming call alerts. You can disable or enable the call alert. You can also activate or deactivate the caller ID display. See Actionable Incoming Call Alert, Product Specific Configuration, on page 136.
AES 256 Encryption Support for Phones	Enhances security by supporting TLS 1.2 and new ciphers. For more information, see Supported Security Features, on page 82.

Feature	Description and more information
Agent Greeting	Allows an agent to create and update a prerecorded greeting that plays at the beginning of a customer call, before the agent begins the conversation with the caller. The agent can prerecord a single greeting or multiple ones as needed. See Enable Agent Greeting, on page 163.
Always On Mode	Always keeps the DECT connection between the headset and base even when the user is not on a call or playing music. This feature is supported on Cisco Headset 500 Series. See Headset template management in Call Manager for more information.
Any Call Pickup	Allows users to pick up a call on any line in their call pickup group, regardless of how the call was routed to the phone. See call pickup in the documentation for your particular Cisco Unified Communications Manager release.
Application Dial Rules	Convert numbers for shared mobile contacts to network dialable numbers. See Application Dial Rules, on page 75.
Assisted Directed Call Park	Enables users to park a call by pressing only one button using the Direct Park feature. Administrators must configure a Busy Lamp Field (BLF) Assisted Directed Call Park button. When users press an idle BLF Assisted Directed Call Park button for an active call, the active call is parked at the Direct Park slot associated with the Assisted Directed Call Park button. See assisted directed call park in the documentation for your particular Cisco Unified Communications Manager release.
Audible Message Waiting Indicator (AMWI)	A stutter tone from the handset, headset, or speakerphone indicates that a user has one or more new voice messages on a line. **Note** The stutter tone is line-specific. You hear it only when using the line with the waiting messages.
Auto Answer	Connects incoming calls automatically after a ring or two. Auto Answer works with either the speakerphone or the headset. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Automatic Port Synchronization	Synchronizes ports to the lowest speed between ports of a phone to eliminate packet loss. See Automatic Port Synchronization, Product Specific Configuration, on page 136.
Auto Pickup	Allows a user to use one-touch pickup functionality for call pickup features. See call pickup in the documentation for your particular Cisco Unified Communications Manager release.

Feature	Description and more information
Barge	Enables a user to barge into a call by establishing the three-way conference call using the built-in conference bridge of the target phone. See “cBarge” in this table.
Block External to External Transfer	Prevents users from transferring an external call to another external number. See external call transfer information in the documentation for your particular Cisco Unified Communications Manager release.
Bluetooth Multiconnection	Enables the user to pair multiple devices to the phone. The user can then connect a mobile device using Bluetooth and a Bluetooth headset at the same time. Cisco IP Phone 8851NR does not support Bluetooth.
Bluetooth Mute Sync for Cisco Headset 700 Series	The mute status between the IP Phone and Cisco headset is synchronized automatically during a call. Supported on the Cisco IP Phones 8845, 8851, 8861, and 8865, connected with Cisco Headset 730 or Cisco Head 720 Series.
Busy Lamp Field (BLF)	Allows a user to monitor the call state of a directory number associated with a speed-dial button on the phone. See presence information in the documentation for your particular Cisco Unified Communications Manager release.
Busy Lamp Field (BLF) Pickup	Provides enhancements to BLF speed dial. Allows you to configure a Directory Number (DN) that a user can monitor for incoming calls. When the DN receives an incoming call, the system alerts the monitoring user, who can then pick up the call. See call pickup information in the documentation for your particular Cisco Unified Communications Manager release.
Call Back	Provides users with an audio and visual alert on the phone when a busy or unavailable party becomes available. See call back information in the documentation for your particular Cisco Unified Communications Manager release.
Call Display Restrictions	Determines the information that will display for calling or connected lines, depending on the parties who are involved in the call. See routing plans and call display restriction information in the documentation for your particular Cisco Unified Communications Manager release.
Call Forward	Allows users to redirect incoming calls to another number. Call Forward options include Call Forward All, Call Forward Busy, Call Forward No Answer, and Call Forward No Coverage. See directory number information in the documentation for your particular Cisco Unified Communications Manager release and Customize the Self Care Portal Display, on page 78.
Call Forward All Loop Breakout	Detects and prevents Call Forward All loops. When a Call Forward All loop is detected, the Call Forward All configuration is ignored and the call rings through.

Feature	Description and more information
Call Forward All Loop Prevention	Detects and prevents Call Forward All loops. When a Call Forward All loop is detected, the Call Forward All configuration is ignored and the call rings through.
Call Forward Configurable Display	Prevents a user from configuring a Call Forward All destination directly on the phone that creates a Call Forward All loop or that creates a Call Forward All chain with more hops than the existing Forward Maximum Hop Count service parameter allows. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Call Forward Destination Override	Allows you to override Call Forward All (CFA) in cases where the CFA target places a call to the CFA initiator. This feature allows the CFA target to reach the CFA initiator for important calls. The override works whether the CFA target phone number is internal or external. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Call Forward Notification	Allows you to configure the information that the user sees when receiving a forwarded call. See Set Up Call Forward Notification, on page 164.
Call History for Shared Line	Allows you to view shared line activity in the phone Call History. This feature will: • Log missed calls for a shared line • Log all answered and placed calls for a shared line
Call Park	Allows users to park (temporarily store) a call and then retrieve the call by using another phone in the Cisco Unified Communications Manager system. You can configure the field **Dedicate one line for call park** in the **Product Specific Configuration Layout** pane to park the call to the original line or a different line. When the field is enabled, the parked call remains on the user's line and they can use the **Resume** softkey to pick up the call. The user sees the extension number for the parked call on the phone display. When the field is disabled, the parked call transfers to the call park line. The user's line returns to the idle state and they see the call park extension in a pop-up window. The user dials the extension to pick up the call. See call park information in the documentation for your particular Cisco Unified Communications Manager release.
Call Pickup	Allows users to redirect a call that is ringing on another phone within their pickup group to their phone. You can configure an audio and visual alert for the primary line on the phone. This alert notifies the users that a call is ringing in their pickup group. See call pickup information in the documentation for your particular Cisco Unified Communications Manager release.

Feature	Description and more information
Call Recording	Allows a supervisor to record an active call. The user might hear a recording audible alert tone during a call when it is being recorded. When a call is secured, the security status of the call is displayed as a lock icon on Cisco IP Phones. The connected parties might also hear an audible alert tone that indicates the call is secured and is being recorded. **Note** When an active call is being monitored or recorded, the user can receive or place intercom calls; however, if the user places an intercom call, the active call is put on hold, which causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the party whose call is being monitored must resume the call. See monitoring and recording information in the documentation for your particular Cisco Unified Communications Manager release.
Call Waiting	Indicates (and allows users to answer) an incoming call that rings while on another call. Incoming call information appears on the phone display. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Call Waiting Ring	Provides Call Waiting users with the option of an audible ring instead of the standard beep. Options are Ring and Ring Once. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Caller ID	Caller identification such as a phone number, name, or other descriptive text appear on the phone display. See routing plan, call display restrictions, and directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Caller ID Blocking	Allows a user to block their phone number or email address from phones that have caller identification enabled. See routing plan and directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Calling Party Normalization	Calling party normalization presents phone calls to the user with a dialable phone number. Any escape codes are added to the number so that the user can easily connect to the caller again. The dialable number is saved in the call history and can be saved in the Personal Address Book.
CAST for SIP	Establishes communication between the Cisco Unified Video Advantage (CUVA) and the Cisco IP phones to support video on the PC even if the IP phone does not have video capability.

Feature	Description and more information
cBarge	Allows a user to join a nonprivate call on a shared phone line. cBarge adds a user to a call and converts it into a conference, allowing the user and other parties to access conference features. The conference call is created using the Cisco Unified Communications Manager conference bridge functionality. You must enable both the softkey and the conference bridge functionality for cBarge to function correctly. In Firmware Release 10.2(2) and later, the cBarge functionality is accessed using the Barge softkey. For more information, refer to the "Barge" chapter, Feature Configuration Guide for Cisco Unified Communications Manager.
Charge a Mobile Device	Allows a user to charge a mobile device by connecting it to the USB port of the Cisco IP Phone. See *Cisco IP Phone 8800 Series User Guide*.
Cisco Extension Mobility	Allows users access to their Cisco IP Phone configuration such as line appearances, services, and speed dials from a shared Cisco IP Phone. Cisco Extension Mobility is useful if people work from a variety of locations within your company or if they share a workspace with coworkers.
Cisco Extension Mobility Cross Cluster (EMCC)	Enables a user configured in one cluster to log into a Cisco IP Phone in another cluster. Users from a home cluster log into a Cisco IP Phone at a visiting cluster. **Note** Configure Cisco Extension Mobility on Cisco IP Phones before you configure EMCC.
Cisco IP Manager Assistant (IPMA)	Provides call routing and other call management features to help managers and assistants handle phone calls more effectively. See Set up Cisco IP Manager Assistant, on page 178.
Cisco IP Phone 8800 Key Expansion Module Cisco IP Phone 8851/8861 Key Expansion Module Cisco IP Phone 8865 Key Expansion Module	Provides additional keys by adding an expansion module to the phone. For additional information, see *Cisco IP Phone 7800 and 8800 Series Accessories Guide for Cisco Unified Communications Manager*.
Cisco IP Phone 8811 Support	Provides support for the Cisco IP Phone 8811.
Cisco IP Phone 8851NR Support	Provides support for the Cisco IP Phone 8851NR

Feature	Description and more information
Cisco Unified Communications Manager Express (Unified CME) Version Negotiation	The Cisco Unified Communication Manager Express uses a special tag in the information sent to the phone to identify itself. This tag enables the phone to provide services to the user that the switch supports. See: • *Cisco Unified Communications Manager Express System Administrator Guide* • Cisco Unified Communications Manager Express Interaction, on page 20
Cisco Unified Video Advantage (CUVA)	Allows users to make video calls by using a Cisco IP Phone, a personal computer, and a video camera. **Note** Configure the Video Capabilities parameter in the Product Specific Configuration Layout section in Phone Configuration. See the Cisco Unified Video Advantage documentation.
Cisco WebDialer	Allows users to make calls from web and desktop applications.
Classic Ringtone	Supports ringtones that are embedded in the phone firmware or downloaded from the Cisco Unified Communications Manager. The feature makes the available ringtones common with other Cisco IP Phones. See Custom Phone Rings, on page 109.
Conference	Allows a user to talk simultaneously with multiple parties by calling each participant individually. Conference features include Conference and Meet Me. Allows a noninitiator in a standard (adhoc) conference to add or remove participants; also allows any conference participant to join together two standard conferences on the same line. The Advance Adhoc Conference service parameter, disabled by default in Cisco Unified Communications Manager Administration, allows you to enable these features. **Note** Be sure to inform your users if these features are activated.
Configurable Energy Efficient Ethernet (EEE) for PC and Switch Port	Provides a method to control EEE functions on personal computer port and switch port by enabling or disabling EEE. The feature controls both type of ports individually. The default value is Enabled. See Set Up Energy Efficient Ethernet for Switch and PC Port, on page 166.
Configurable Font Size	Allows users to increase or decrease the maximum number of characters the IP phone displays for Call History and Call Screen by changing the font size. A smaller font increases the maximum number of displayed characters, and a larger font decreases the maximum number of displayed characters.
CTI Applications	A computer telephony integration (CTI) route point can designate a virtual device to receive multiple, simultaneous calls for application-controlled redirection.

Feature	Description and more information
Decline All	Allows a user to transfer a ringing, connected, or held call directly to a voice-messaging system. When a call is declined, the line becomes available to make or receive new calls. See immediate divert information in the documentation for your particular Cisco Unified Communications Manager release.
Device Invoked Recording	Provides end users with the ability to record their telephone calls via a softkey. In addition administrators may continue to record telephone calls via the CTI User Interface. See monitoring and recording information in the documentation for your particular Cisco Unified Communications Manager release.
Directed Call Park	Allows a user to transfer an active call to an available directed call park number that the user dials or speed dials. A Call Park BLF button indicates whether a directed call park number is occupied and provides speed-dial access to the directed call park number. **Note** If you implement Directed Call Park, avoid configuring the Park softkey. This prevents users from confusing the two Call Park features. See call park information in the documentation for your particular Cisco Unified Communications Manager release.
Display Battery Strength and Signal Strength Icons	Displays the battery and signal strength of the mobile phone on the IP Phone when the mobile phone is connected to the IP Phone using Bluetooth. Cisco IP Phone 8851NR does not support Bluetooth.
Distinctive Ring	Users can customize how their phone indicates an incoming call and a new voice mail message. See call pickup information in the documentation for your particular Cisco Unified Communications Manager release.
Do Not Disturb (DND)	When DND is turned on, either no audible rings occur during the ringing-in state of a call, or no audible or visual notifications of any type occur. When enabled, the phone header turns red and Do not disturb is displayed on the phone. If multilevel precedence and preemption (MLPP) is configured and the user receives a precedence call, the phone will ring with a special ringtone. See Set Up Do Not Disturb, on page 162.
Dock Event	Allows a user to set up the call behavior when the headset is lift from the base or is put down on the base. This feature is supported on Cisco Headset 500 Series. See Headset template management in Call Manager for more information.
Enable/Disable JAL/TAL	Allows the administrator to control the Join Across Lines (JAL) and Direct Transfer Across Lines (TAL) features. See Join and Direct Transfer Policy, Product Specific Configuration, on page 136.

Feature	Description and more information
EnergyWise	Enables an IP Phone to sleep (power down) and wake (power up) at predetermined times, to promote energy savings. See Schedule EnergyWise on Cisco IP Phone, on page 158.
Enhanced Line Mode	Enable Enhanced Line Mode to use the buttons on both sides of the phone screen as line keys. See Set Up Additional Line Keys, on page 189
Enhanced Secure Extension Mobility Cross Cluster (EMCC)	Improves the Secure Extension Mobility Cross Cluster (EMCC) feature by preserving the network and security configurations on the login phone. By so doing, security policies are maintained, network bandwidth is preserved and network failure is avoided within the visiting cluster (VC).
Fast Dial Service	Allows a user to enter a Fast Dial code to place a call. Fast Dial codes can be assigned to phone numbers or Personal Address Book entries. See “Services” in this table. See Modify Phone Button Template for PAB or Fast Dial, on page 188.
Group Call Pickup	Allows a user to answer a call that is ringing on a directory number in another group. See call pickup information in the documentation for your particular Cisco Unified Communications Manager release.
Headset Sidetone Control	Allows an administrator to set the sidetone level of a wired headset.
Hold Reversion	Limits the amount of time that a call can be on hold before reverting back to the phone that put the call on hold and alerting the user. Reverting calls are distinguished from incoming calls by a single ring (or beep, depending on the new call indicator setting for the line). This notification repeats at intervals if not resumed. A call that triggers Hold Reversion also displays an animated icon in the call bubble. You can configure call focus priority to favor incoming or reverting calls.
Hold Status	Enables phones with a shared line to distinguish between the local and remote lines that placed a call on hold.
Hold/Resume	Allows the user to move a connected call from an active state to a held state. • No configuration required unless you want to use Music On Hold. See “Music On Hold” in this table for information. • See “Hold Reversion” in this table.
HTTP Download	Enhances the file download process to the phone to use HTTP by default. If the HTTP download fails, the phone reverts to using the TFTP download.

Feature	Description and more information
Hunt Group	Provides load sharing for calls to a main directory number. A hunt group contains a series of directory numbers that can answer the incoming calls. When the first directory number in the hunt group is busy, the system hunts in a predetermined sequence for the next available directory number in the group and directs the call to that phone. You can have **Caller ID** (If the Caller ID is configured), **Directory Number** and **Hunt group Pilot Number** display on the Incoming Call Alert for the hunt group call. The hunt group number is displayed after the label "Hunt Group. See hunt group and routing plans information in the documentation for your particular Cisco Unified Communications Manager release.
Incoming Call Toast Timer	Allows you to set the length of time that an incoming call toast (notification) appears on the phone screen. See Incoming Call Toast Timer, Product Specific Configuration, on page 136.
Intelligent Proximity	Enables users to pair a mobile device with the phone using Bluetooth and use the phone to place and receive mobile calls, See Enable Intelligent Proximity, on page 193. Cisco IP Phone 8811, 8841, and 8851NR do not support Bluetooth or Intelligent Proximity.
Intercom	Allows users to place and receive intercom calls using programmable phone buttons. You can configure intercom line buttons to: • Directly dial a specific intercom extension. • Initiate an intercom call and then prompt the user to enter a valid intercom number. **Note** If your user logs into the same phone on a daily basis using their Cisco Extension Mobility profile, assign the phone button template that contains intercom information to their profile, and assign the phone as the default intercom device for the intercom line.
IPv6-only Support	Provides support for expanded IP addressing on Cisco IP Phones. IPv4 and IPv6 configuration is recommended and fully supported. Certain features are not supported in a standalone configuration. Only IPv6 address are assigned. See Configure Network Settings, on page 55.
Jitter Buffer	The Jitter Buffer feature handles jitter from 10 milliseconds (ms) to 1000 ms for audio streams. It runs in an adaptive mode and it dynamically adjusts to the amount of jitter.
Join	Allows users to combine two calls that are on one line to create a conference call and remain on the call.

Feature	Description and more information
Line Status for Call Lists	Allows the user to see the Line Status availability status of monitored line numbers in the Call History list. The Line Status states are • Offline • Available • In use • Do not disturb See Enable BLF for Call Lists, on page 165.
Line Status in Corporate Directory	Enables the display of the status of a contact in the Corporate Directory. • Offline • Available • In use • Do not disturb See Enable BLF for Call Lists, on page 165.
Line Text Label	Sets a text label for a phone line instead of the directory number. See Set the Label for a Line, on page 174.
Log out of hunt groups	Allows users to log out of a hunt group and temporarily block calls from ringing their phone when they are not available to take calls. Logging out of hunt groups does not prevent nonhunt group calls from ringing their phone. See routing plan information in the documentation for your particular Cisco Unified Communications Manager release.
Malicious Caller Identification (MCID)	Allows users to notify the system administrator about suspicious calls that are received.
Meet Me Conference	Allows a user to host a Meet Me conference in which other participants call a predetermined number at a scheduled time.
Message Waiting	Defines directory numbers for message waiting on and off indicators. A directly-connected voice-message system uses the specified directory number to set or to clear a message waiting indication for a particular Cisco IP Phone. See message waiting and voice mail information in the documentation for your particular Cisco Unified Communications Manager release.
Message Waiting Indicator	A light on the handset that indicates that a user has one or more new voice messages. See message waiting and voice mail information in the documentation for your particular Cisco Unified Communications Manager release.
Minimum Ring Volume	Sets a minimum ringer volume level for an IP phone.

Feature	Description and more information
Missed Call Logging	Allows a user to specify whether missed calls will be logged in the missed calls directory for a given line appearance. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Mobile Connect	Enables users to manage business calls using a single phone number and pick up in-progress calls on the desk phone and a remote device such as a mobile phone. Users can restrict the group of callers according to phone number and time of day. See Cisco Unified Mobility information in the documentation for your particular Cisco Unified Communications Manager release.
Mobile and Remote Access Through Expressway	Allows remote workers to easily and securely connect into the corporate network without using a virtual private network (VPN) client tunnel. See Mobile and Remote Access Through Expressway, on page 167
Mobile Voice Access	Extends Mobile Connect capabilities by allowing users to access an interactive voice response (IVR) system to originate a call from a remote device such as a cellular phone. See Cisco Unified Mobility in the documentation for your particular Cisco Unified Communications Manager release.
Monitoring and Recording	Allows a supervisor to silently monitor an active call. The supervisor cannot be heard by either party on the call. The user might hear a monitoring audible alert tone during a call when it is being monitored. When a call is secured, the security status of the call is displayed as a lock icon on Cisco IP Phones. The connected parties might also hear an audible alert tone that indicates the call is secured and is being monitored. **Note** When an active call is being monitored or recorded, the use can receive or place intercom calls; however, if the user place an intercom call, the active call will be put on hold, which causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the party whose call is being monitored must resume the call.
Multilevel Precedence and Preemption	Enables the user to make and receive urgent or critical calls in some specialized environments, such as military or government offices. See Multilevel Precedence and Preemption, on page 183.
Multiple Calls Per Line Appearance	Each line can support multiple calls. By default, the phone supports two active calls per line, and a maximum of six active calls per line. Only one call can be connected at any time; other calls are automatically placed on hold. The system allows you to configure maximum calls/busy trigger not more than 6/6. Any configuration more than 6/6 is not officially supported. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.
Music On Hold	Plays music while callers are on hold.

Feature	Description and more information
Mute	Mutes the handset or headset microphone.
No Alert Name	Makes it easier for end users to identify transferred calls by displaying the original caller's phone number. The call appears as an Alert Call followed by the caller's telephone number.
Onhook Dialing	Allows a user to dial a number without going off hook. The user can then either pick up the handset or press Dial.
Other Group Pickup	Allows a user to answer a call ringing on a phone in another group that is associated with the user's group. See call pickup information in the documentation for your particular Cisco Unified Communications Manager release.
Phone Display Message for Extension Mobility Users	This feature enhances the phone interface for the Extension Mobility user by providing friendly messages.
Phone Trust List Notification in Cisco Unified Communications Manager	Enables the phone to send an alarm to the Cisco Unified Communications Manager when the Trust List (TL) is updated. See Supported Security Features, on page 82.
PLK Support for Queue Statistics	The PLK Support for Queue Statistics feature enables the users to query the call queue statistics for hunt pilots and the information appears on phone screen.
Plus Dialing	Allows the user to dial E.164 numbers prefixed with a plus (+) sign. To dial the + sign, the user needs to press and hold the star (*) key for at least 1 second. This applies to dialing the first digit for an on-hook (including edit mode) or off-hook call.
Power Negotiation over LLDP	Allows the phone to negotiate power using Link Level Endpoint Discovery Protocol (LLDP) and Cisco Discovery Protocol (CDP). See Power Negotiation, Product Specific Configuration, on page 136.
Predictive Dialing	Simplifies making a call. The Recents list changes to displays only phone numbers similar to the number being dialed. Predictive Dialing is enabled when Enhanced Line mode is enabled. Simplified New Call UI must be disabled for Predictive Dialing to function.
Privacy	Prevents users who share a line from adding themselves to a call and from viewing information on their phone display about the call of the other user. See barge and privacy information in the documentation for your particular Cisco Unified Communications Manager release.

Feature	Description and more information
Private Line Automated Ringdown (PLAR)	The Cisco Unified Communications Manager administrator can configure a phone number that the Cisco IP Phone dials as soon as the handset goes off hook. This can be useful for phones that are designated for calling emergency or “hotline” numbers. The administrator can configure a delay of up to 15-seconds. This allows the user time to place a call before the phone defaults to the hotline number. The timer is configurable through the parameter **Off Hook To First Digit Timer** under **Device** > **Device Settings** > **SIP Profile**. For more information, refer to *Feature Configuration Guide for Cisco Unified Communications Manager*.
Problem Report Tool (PRT)	Submit phone logs or report problems to an administrator. See Problem Report Tool, on page 172.
Programmable Feature Buttons	You can assign features, such as New Call, Call Back, and Forward All to line buttons. See phone button template information in the documentation for your particular Cisco Unified Communications Manager release.
Quality Reporting Tool (QRT)	Allows users to submit information about problem phone calls by pressing a button. QRT can be configured for either of two user modes, depending upon the amount of user interaction desired with QRT.
Recents	Allows users to see the 150 most recent individual calls and call groups. You can see the recently dialed numbers, missed calls, and delete a call record.
Redial	Allows users to call the most recently dialed phone number by pressing a button or the Redial softkey.
Remote Port Configuration	Allows you to configure the speed and duplex function of the phone Ethernet ports remotely by using Cisco Unified Communications Manager Administration. This enhances the performance for large deployments with specific port settings. **Note** If the ports are configured for Remote Port Configuration in Cisco Unified Communications Manager, the data cannot be changed on the phone. See Remote Port Configuration, Product Specific Configuration, on page 136.
Reroute Direct Calls to Remote Destination to Enterprise Number	Reroutes a direct call to a user's mobile phone to the enterprise number (desk phone). For an incoming call to remote destination (mobile phone), only remote destination rings; desk phone does not ring. When the call is answered on their mobile phone, the desk phone displays a Remote In Use message. During these calls, users can make use of various features of their mobile phone. See the Cisco Unified Mobility information in the documentation for your particular Cisco Unified Communications Manager release.
Remove 'Call Ended' Prompt Timer	Improves the End Call response time by removing the `Call ended` message display on the phone screen.

Feature	Description and more information
Ringtone Setting	Identifies ring type used for a line when a phone has another active call. See directory number information in the documentation for your particular Cisco Unified Communications Manager release and Custom Phone Rings, on page 109.
RTCP Hold For SIP	Ensures that held calls are not dropped by the gateway. The gateway checks the status of the RTCP port to determine if a call is active or not. By keeping the phone port open, the gateway will not end held calls.
Secure Conference	Allows secure phones to place conference calls using a secured conference bridge. As new participants are added by using Confrn, Join, or Barge softkeys or MeetMe conferencing, the secure call icon displays as long as all participants use secure phones. The Conference List displays the security level of each conference participant. Initiators can remove nonsecure participants from the Conference List. Noninitiators can add or remove conference participants if the Advanced Adhoc Conference Enabled parameter is set. See conference bridge and security information in the documentation for your particular Cisco Unified Communications Manager release and Supported Security Features, on page 82.
Secure EMCC	Improves the EMCC feature by providing enhanced security for a user logging into their phone from a remote office.
Services	Allows you to use the Cisco IP Phone Services Configuration menu in Cisco Unified Communications Manager Administration to define and maintain the list of phone services to which users can subscribe. See services information in the documentation for your particular Cisco Unified Communications Manager release.
Services URL button	Allows users to access services from a programmable button rather than by using the Services menu on a phone. See services information the documentation for your particular Cisco Unified Communications Manager release.
Show Calling ID and Calling Number	The phones can display both the calling ID and calling number for incoming calls. The IP phone LCD display size limits the length of the calling ID and the calling number that display. The Show Calling ID and Calling Number feature applies to the incoming call alert only and does not change the function of the Call Forward and Hunt Group features. See “Caller ID” in this table.

Feature	Description and more information
Simplify Extension Mobility Login with Cisco Headsets	Enables users to sign into Extension Mobility with their Cisco headsets. When the phone is in MRA mode, the user can use the headset to sign into the phone. This feature requires Cisco Unified Communications Manager(UCM) Release 11.5(1)SU8, 11.5(1)SU.9, 12.5(1)SU3 or later. For more information, see *Feature Configuration Guide for Cisco Unified Communications Manager*, Release 11.5(1)SU8 or later, or Release 12.5(1)SU3 or later. With Cisco Headset 730, user can also sign into Extension Mobility with the headset USB adapter. The required version of the headset is 1-8-0-213 or later, and the USB adapter is 1-3-20 or later.
Simplified Tablet Support	Enables an Android or iOS tablet user to pair the tablet to the phone using Bluetooth and then use the phone for the audio part of a call on the tablet. See Enable Intelligent Proximity, on page 193. Cisco IP Phone 8851NR does not support Bluetooth.
Speed Dial	Dials a specified number that has been previously stored.
SSH Access	Allows you to enable or disable the SSH Access setting using Cisco Unified Communications Manager Administration. Enabling the SSH server allows the phone to accept the SSH connections. Disabling the SSH server functionality of the phone blocks the SSH access to the phone. See SSH Access, Product Specific Configuration, on page 136.
Time-of-Day Routing	Restricts access to specified telephony features by time period. See time period and time-of-day routing information in the documentation for your particular Cisco Unified Communications Manager release.
Time Zone Update	Updates the Cisco IP Phone with time zone changes. See date and time information in the documentation for your particular Cisco Unified Communications Manager release.
Transfer	Allows users to redirect connected calls from their phones to another number.
Transfer - Direct Transfer	Transfer: The first invocation of Transfer will always initiate a new call by using the same directory number, after putting the active call on hold. The user can directly transfer calls using Transfer Active Call Function. Some JTAPI/TAPI applications are not compatible with the Join and Direct Transfer feature implementation on the Cisco IP Phone and you may need to configure the Join and Direct Transfer Policy to disable join and direct transfer on the same line or possibly across lines. See directory number information in the documentation for your particular Cisco Unified Communications Manager release.

Feature	Description and more information
TVS	Trust Verification Services (TVS) enables phones to authenticate signed configurations and authenticate other servers or peers without increasing the size of the Certificate Trust List (CTL) or requiring the downloading of an updated CTL file to the phone. TVS is enabled by default. The Security Setting menu on the phone displays the TVS information.
UCR 2013	The Cisco IP Phones support Unified Capabilities Requirements (UCR) 2013 by providing the following functions: • Support for Federal Information Processing Standard (FIPS) 140-2 • Support for 80-bit SRTCP Tagging As an IP Phone administrator, you must set up specific parameters in Cisco Unified Communications Manager Administration.
Unconfigured Primary Line Notification	Alerts the user when the primary line is not configured. The user sees the message `Unprovisioned` on the phone screen.
User Interface Updates for List, Alert, and Visual Voicemail.	Increases the size of the application window to minimize truncated strings.
Video Mode	Allows a user to select the video display mode for viewing a video conference, depending on the modes configured in the system. See video information in the documentation for your particular Cisco Unified Communications Manager release. Available on Cisco IP Phone 8845, 8865, and 8865NR.
Video Support	Enables video support on the phone. The Video Capabilities parameter needs to be enabled for video calls on Cisco Unified Communications Manager Phone Configuration window. It is enabled by default. Available on Cisco IP Phone 8845, 8865, and 8865NR.
Video Through PC	Allows users to make video calls by using their Cisco Unified IP Phone, personal computer, and an external video camera. The feature also allows users to make video calls with Cisco Jabber or Cisco Unified Video Advantage products.
Visual Voicemail	Replaces the voicemail audio prompts with a graphical interface. See *Installation and Configuration Guide for Visual Voicemail* located at http://www.cisco.com/en/US/partner/products/ps9829/prod_installation_guides_list.html#anchor3.
Voice Message System	Enables callers to leave messages if calls are unanswered. See voice mail information in the documentation for your particular Cisco Unified Communications Manager release and Set up Visual Voicemail, on page 180.

Feature	Description and more information
VPN	Using SSL, provides a virtual private network (VPN) connection on the Cisco Unified IP Phone when it is located outside a trusted network or when network traffic between the phone and Unified Communications Manager must cross untrusted networks.
Web Access Disabled by Default	Enhances security by disabling access to all web services, such as HTTP. Users can only access web services if you enable web access.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Feature Buttons and Softkeys

The following table provides information about features that are available on softkeys, features that are available on dedicated feature buttons, and features that you need to configure as programmable feature buttons. A "Supported" entry in the table indicates that the feature is supported for the corresponding button type or softkey. Of the two button types and softkeys, only programmable feature buttons require configuration in Cisco IP Phone administration.

For information about configuring programmable feature buttons, see Phone Button Templates, on page 185.

Table 30: Features and Corresponding Buttons and Softkeys

Feature name	Dedicated feature button	Programmable feature button	Softkey
Alert Calls	Not supported	Supported	Not supported
All Calls	Not supported	Supported	Not supported
Answer	Not supported	Supported	Supported
cBarge	Not supported	Not supported	Supported
Call Back	Not supported	Supported	Supported
Call Forward All	Not supported	Not supported	Supported
Call Park	Not supported	Supported	Supported
Call Park Line Status	Not supported	Supported	Not supported
Call Pickup (Pick Up)	Not supported	Supported	Supported
Call Pickup Line Status	Not supported	Supported	Not supported
Conference	Supported	Not supported	Supported
Divert	Not supported	Not supported	Supported
Do Not Disturb	Not supported	Supported	Supported

Feature name	Dedicated feature button	Programmable feature button	Softkey
Group Pickup (Group Pick Up)	Not supported	Supported	Supported
Hold	Supported	Not supported	Supported
Hunt Groups	Not supported	Supported	Not supported
Intercom	Not supported	Supported	Not supported
Malicious Call Identification (MCID)	Not supported	Supported	Supported
Meet Me	Not supported	Supported	Supported
Merge	Not supported	Not supported	Supported
Mobile Connect (Mobility)	Not supported	Supported	Supported
Mute	Supported	Not supported	Not supported
Other Pickup	Not supported	Supported	Supported
PLK Support for Queue Status	Not supported	Not supported	Supported
Privacy	Not supported	Supported	Not supported
Queue Status	Not supported	Supported	Not supported
Quality Reporting Tool (QRT)	Not supported	Supported	Supported
Record	Not supported	Not supported	Supported
Redial	Not supported	Supported	Supported
Speed Dial	Not supported	Supported	Not supported
Speed Dial Line Status	Not supported	Supported	Not supported
Support for Hold Button on USB Headsets	Not supported	Not supported	Supported
Transfer	Supported	Not supported	Supported

Phone Feature Configuration

You can set up phones to have a variety of features, based on the needs of your users. You can apply features to all phones, a group of phones, or to individual phones.

When you set up features, the Cisco Unified Communications Manager Administration window displays information that is applicable to all phones and information that is applicable to the phone model. The information that is specific to the phone model is in the Product Specific Configuration Layout area of the window.

For information on the fields applicable to all phone models, see the Cisco Unified Communications Manager documentation.

When you set a field, the window that you set the field in is important because there is a precedence to the windows. The precedence order is:

1. Individual phones (highest precedence)
2. Group of phones
3. All phones (lowest precedence)

For example, if you don't want a specific set of users to access the phone Web pages, but the rest of your users can access the pages, you:

1. Enable access to the phone web pages for all users.
2. Disable access to the phone web pages for each individual user, or set up a user group and disable access to the phone web pages for the group of users.
3. If a specific user in the user group did need access to the phone web pages, you could enable it for that particular user.

Set Up Phone Features for All Phones

Procedure

Step 1 Sign in to Cisco Unified Communications Manager Administration as an administrator.

Step 2 Select **System** > **Enterprise Phone Configuration**.

Step 3 Set the fields you want to change.

Step 4 Check the **Override Enterprise Settings** check box for any changed fields.

Step 5 Click **Save**.

Step 6 Click **Apply Config**.

Step 7 Restart the phones.

Note This will impact all phones in your organization.

Set Up Phone Features for a Group of Phones

Procedure

Step 1 Sign in to Cisco Unified Communications Manager Administration as an administrator.
Step 2 Select **Device** > **Device Settings** > **Common Phone Profile**.
Step 3 Locate the profile.
Step 4 Navigate to the Product Specific Configuration Layout pane and set the fields.
Step 5 Check the **Override Enterprise Settings** check box for any changed fields.
Step 6 Click **Save**.
Step 7 Click **Apply Config**.
Step 8 Restart the phones.

Set Up Phone Features for a Single Phone

Procedure

Step 1 Sign in to Cisco Unified Communications Manager Administration as an administrator.
Step 2 Select **Device** > **Phone**
Step 3 Locate the phone associated with the user.
Step 4 Navigate to the Product Specific Configuration Layout pane and set the fields.
Step 5 Check the **Override Common Settings** check box for any changed fields.
Step 6 Click **Save**.
Step 7 Click **Apply Config**.
Step 8 Restart the phone.

Product Specific Configuration

The following table describes the fields in the Product Specific Configuration Layout pane.

Table 31: Product Specific Configuration Fields

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Disable Speakerphone	Check box	Unchecked	Turns off the speakerphone capability of the phone.
Disable Speakerphone and Headset	Check box	Unchecked	Turns off the speakerphone and headset capability of the phone.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Disable Handset	Checkbox	Unchecked	Turns off the handset capability of the phone.
PC Port	Enabled Disabled	Enabled	Controls the ability to use the PC port to connect a computer into the LAN.
Settings Access	Disabled Enabled Restricted	Enabled	Enables, disables, or restricts access to the local phone configuration settings in the Settings app. • Disabled—The Settings menu does not display any options. • Enabled—All entries in the Settings menu are accessible. • Restricted—Only the Phone settings menu is accessible.
PC Voice VLAN Access	Enabled Disabled	Enabled	Indicates whether the phone will allow a device attached to the PC port to access the Voice VLAN. • Disabled—The PC can't send and receive data on the Voice VLAN or from the phone. • Enabled—The PC can send and receive data from the Voice VLAN or from the phone. Set this field to Enabled if an application is being run on the PC that to monitor phone traffic. These applications could include monitoring and recording applications, and the use of network monitoring software for analysis purposes.
Video Capabilities	Enabled Disabled	8845, 8865, and 8865NR: Enabled 8811, 8851, 8851NR, 8861: Disabled	Allows users to make video calls by using a Cisco IP Phone, a personal computer, and a video camera.
Web Access	Disabled Enabled	Disabled	Enables or disables access to the phone web pages through a web browser. **Caution** If you enable this field, you may expose sensitive information about the phone.
Disable TLS 1.0 and TLS 1.1 for Web Access	Disabled Enabled	Disabled	Controls the use of TLS 1.2 for a web server connection. • Disabled—A phone configured for TLS1.0, TLS 1.1, or TLS1.2 can function as a HTTPs server. • Enabled—Only a phone configured for TLS1.2 can function as a HTTPs server.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Enbloc Dialing	Disabled Enabled	Disabled	Controls the dialing method. • Disabled—The Cisco Unified Communications Manager waits for the interdigit timer to expire when there is a dial plan or route pattern overlap. • Enabled—The entire dialed string is sent to Cisco Unified Communications Manager once the dialing is complete. To avoid the T.302 timer timeout, we recommend that you enable Enbloc Dialing whenever there is a dialplan or route pattern overlap. Forced Authorization Codes (FAC) or Client Matter Codes (CMC) do not support the Enbloc Dialing. If you use FAC or CMC to manage call access and accounting, then you cannot use this feature.
Days Display Not Active	Days of the week		Defines the days that the display does not turn on automatically at the time specified in the Display On Time field. Choose the day or days from the drop-down list. To choose more than one day, **Ctrl+click** each day that you want.
Display On Time	hh:mm		Defines the Time each day that the display turns on automatically (except on the days specified in the Days Display Not Active field). Enter the time in this field in 24 hour format, where 0:00 is midnight. For example, to automatically turn the display on at 07:00 a.m. (0700), enter 07:00. To turn the display on at 02:00 p.m. (1400), enter 14:00. If this field is blank, the display automatically turns on at 0:00.
Display On Duration	hh:mm		Defines the length of time that the display remains on after turning on at the time specified in the Display On Time field. For example, to keep the display on for 4 hours and 30 minutes after it turns on automatically, enter 04:30. If this field is blank, the phone turns off at the end of the day (0:00). If Display On Time is 0:00 and the display on duration is blank (or 24:00), the display does not turn off.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Display Idle Timeout	hh:mm	01:00	Defines the length of time that the phone is idle before the display turns off. Applies only when the display was off as scheduled and was turned on by a user (by pressing a button on the phone or lifting the handset). Enter the value in this field in the format hours:minutes. For example, to turn the display off when the phone is idle for 1 hour and 30 minutes after a user turns the display on, enter 01:30. For more information, see Set Up Idle Display, on page 111.
Display On When Incoming Call	Disabled Enabled	Enabled	Turns the idle display on when there is an incoming call.
Enable Power Save Plus	Days of the week		Defines the schedule of days for which the phone powers off. Choose the day or days from the drop-down list. To choose more than one day, **Ctrl+click** each day that you want. When Enable Power Save Plus is turned on, you receive a message that warns about emergency (e911) concerns. **Caution** While Power Save Plus Mode (the "Mode") is in effect, endpoints that are configured for the mode are disabled for emergency calling and from receiving inbound calls. By selecting this mode, you agree to the following: (i) You take full responsibility for providing alternate methods for emergency calling and receiving calls while the mode is in effect; (ii) Cisco has no liability in connection with your selection of the mode and all liability in connection with enabling the mode is your responsibility; and (iii) You fully inform users of the effects of the mode on calls, calling and otherwise. To disable Power Save Plus, you must uncheck the Allow EnergyWise Overrides check box. If the Allow EnergyWise Overrides remains checked but no days are selected in the Enable Power Save Plus field, Power Save Plus is not disabled.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Phone On Time	hh:mm		Determines when the phone automatically turns on for the days that are in the Enable Power Save Plus field. Enter the time in this field in 24-hour format, where 00:00 is midnight. For example, to automatically power up the phone at 07:00 a.m. (0700), enter 07:00. To power up the phone at 02:00 p.m. (1400), enter 14:00. The default value is blank, which means 00:00. The Phone On Time must be at least 20 minutes later than the Phone Off Time. For example, if the Phone Off Time is 07:00, the Phone On Time must be no earlier than 07:20.
Phone Off Time	hh:mm		Identifies the time of day that the phone powers down for the days that are selected in the Enable Power Save Plus field. If the Phone On Time and the Phone Off Time fields contain the same value, the phone does not power down. Enter the time in this field in 24-hour format, where 00:00 is midnight. For example, to automatically power down the phone at 7:00 a.m. (0700), enter 7:00. To power down the phone at 2:00 p.m. (1400), enter 14:00. The default value is blank, which means 00:00. The Phone On Time must be at least 20 minutes later than the Phone Off Time. For example, if the Phone Off Time is 7:00, the Phone On Time must be no earlier than 7:20.
Phone Off Idle Timeout	20 to 1440 minutes	60	Indicates the length of time that the phone must be idle before the phone powers down. The timeout occurs under the following conditions: • When the phone was in Power Save Plus mode, as scheduled, and was taken out of Power Save Plus mode because the phone user pressed the Select key. • When the phone is repowered by the attached switch. • When the Phone Off Time is reached but the phone is in use.
Enable Audible Alert	Check box	Unchecked	When enabled, instructs the phone to play an audible alert starting 10 minutes before the time that the Phone Off Time field specifies. This check box applies only if the Enable Power Save Plus list box has one or more days selected.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
EnergyWise Domain	Up to 127 characters		Identifies the EnergyWise domain that the phone is in.
EnergyWise Secret	Up to 127 characters		Identifies the security secret password that is used to communicate with the endpoints in the EnergyWise domain.
Allow EnergyWise Overrides	Check box	Unchecked	Determines whether you allow the EnergyWise domain controller policy to send power level updates to the phones. The following conditions apply: • One or more days must be selected in the Enable Power Save Plus field. • The settings in Cisco Unified Communications Manager Administration take effect on schedule even if EnergyWise sends an override. For example, assuming the Phone Off Time is set to 22:00 (10:00 p.m.), the value in the Phone On Time field is 06:00 (6:00 a.m.), and the Enable Power Save Plus has one or more days selected. • If EnergyWise directs the phone to turn off at 20:00 (8:00 p.m.), that directive remains in effect (assuming no phone user intervention occurs) until the configured Phone On Time at 6:00 a.m. • At 6:00 a.m., the phone turns on and resumes receiving the power level changes from the settings in Cisco Unified Communications Manager Administration. • To change the power level on the phone again, EnergyWise must reissue a new power level change command. To disable Power Save Plus, you must uncheck the Allow EnergyWise Overrides check box. If the Allow EnergyWise Overrides remains checked but no days are selected in the Enable Power Save Plus field, Power Save Plus is not disabled.
Join and Direct Transfer Policy	Same line, across line enable Same line enable only Same line, across line disable	Same line, across line enable	Controls the ability of a user to join and transfer calls. • Same line, across line enable—Users can directly transfer or join a call on current line to another call on another line. • Same line enable only—Users can only directly transfer or join the calls when both calls are on same line. • Same line, across line disable— Users can't join or transfer calls on the same line. The join and transfer features are disabled and the user can't do the direct transfer or join function.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Span to PC Port	Disabled Enabled	Disabled	Indicates whether the phone forwards packets that are transmitted and received on the network port to the access port.
Recording Tone	Disabled Enabled	Disabled	Controls the playing of the tone when a user is recording a call.
Recording Tone Local Volume	Integer 0–100	100	Controls the volume of the recording tone to the local user.
Recording Tone Remote Volume	Integer 0–100	50	Controls the volume of the recording tone to the remote user.
Recording Tone Duration	Integer 1–3000 milliseconds		Controls the duration of the recording tone.
Log Server	String of up to 256 characters		Identifies the IPv4 syslog server for phone debug output. The format for the address is: `address:<port>@@base=<0-7>;pfs=<0-1>`
Cisco Discovery Protocol (CDP): Switch Port	Disabled Enabled	Enabled	Controls Cisco Discovery Protocol on the SW port of the phone.
Cisco Discovery Protocol (CDP): PC Port	Disabled Enabled	Enabled	Controls Cisco Discovery Protocol on the PC port of the phone.
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port	Disabled Enabled	Enabled	Enables LLDP-MED on the SW port.
Link Layer Discovery Protocol (LLDP): PC Port	Disabled Enabled	Enabled	Enables LLDP on the PC port.
LLDP Asset ID	String, up to 32 characters		Identifies the asset ID that is assigned to the phone for inventory management.
LLDP Power Priority	Unknown Low High Critical	Unknown	Assigns a phone power priority to the switch, thus enabling the switch to appropriately provide power to the phones.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
802.1x Authentication	User Controlled Enabled Disabled	User Controlled	Specifies the 802.1x authentication feature status. • User Controlled—The user can configure the 802.1x on the phone. • Disabled—802.1x Authentication is not used. • Enabled—802.1x authentication is used, and you configure the authentication for the phones.
Automatic Port Synchronization	Disabled Enabled	Disabled	Synchronizes ports to the lowest speed between ports of a phone to eliminate packet loss.
Switch Port Remote Configuration	Disabled Enabled	Disabled	Allows you to configure the speed and duplex function of the phone SW port remotely. This enhances the performance for large deployments with specific port settings. If the SW ports are configured for Remote Port Configuration in Cisco Unified Communications Manager, the data cannot be changed on the phone.
PC Port Remote Configuration	Disabled Enabled	Disabled	Allows you to configure the speed and duplex function of the phone PC port remotely. This enhances the performance for large deployments with specific port settings. If the ports are configured for Remote Port Configuration in Cisco Unified Communications Manager, the data cannot be changed on the phone.
SSH Access	Disabled Enabled	Disabled	Controls the access to the SSH daemon through port 22. Leaving port 22 open leaves the phone vulnerable to Denial of Service (DoS) attacks.
Incoming Call Toast Timer	0, 3, 4, 5, 6, 7, 8, 9, 10, 15, 30, 60	5	Gives the time, in seconds, that the toast displays. The time includes the fade-in and fade-out times for the window. 0 means that the incoming call toast is disabled.
Ring Locale	Default Japan	Default	Controls the ringing pattern.
TLS Resumption Timer	Integer 0–3600 seconds	3600	Controls the ability to resume a TLS session without repeating the entire TLS authentication process. If the field is set to 0, then the TLS session resumption is disabled.
FIPS Mode	Disabled Enabled	Disabled	Enables or disables the Federal Information Processing Standards (FIPS) mode on the phone.
Record Call Log from Shared Line	Disabled Enabled	Disabled	Specifies whether to record a shared line call in the call log.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Minimum Ring Volume	0-Silent 1–15	0-Silent	Controls the minimum ring volume for the phone. You can set a phone so that the ringer cannot be turned off.
Peer Firmware Sharing	Disabled Enabled	Enabled	Allows the phone to find other phones of the same model on the subnet and share updated firmware files. If the phone has a new firmware load, it can share that load with the other phones. If one of the other phones has a new firmware load, the phone can download the firmware from the other phone, instead of from the TFTP server. Peer firmware sharing: • Limits congestion on TFTP transfers to centralized remove TFTP servers. • Eliminates the need to manually control firmware upgrades. • Reduces phone downtime during upgrades when large numbers of phones are reset simultaneously. • Helps with firmware upgrades in branch or remote office deployment scenarios that run over bandwidth-limited WAN links.
Load Server	String of up to 256 characters		Identifies the alternate IPv4 server that the phone uses to obtain firmware loads and upgrades. The format for the address is: **address:<port>@@base=<0-7>;pfs=<0-1>**
IPv6 Load Server	String of up to 256 characters		Identifies the alternate IPv6 server that the phone uses to obtain firmware loads and upgrades. The format for the address is: **[address]:<port>@@base=<0-7>;pfs=<0-1>**
Wideband Headset UI Control	Disabled Enabled	Enabled	Allows the user to use the wideband codec for an analog headset.
Wideband Headset	Disabled Enabled	Enabled	Enables or disables the use of a Wideband Headset on the phone. Used in conjunction with User Control Wideband Headset. For more information, see Set Up Wideband Codec, on page 111.
Wi-Fi	Disabled Enabled	Enabled	Enables the Cisco IP Phones 8861 and 8865 to connect to the Wi-Fi network. Phones that do not support this feature do not display the field.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Back USB Port	Disabled Enabled	8861, 8865, and 8865NR: Enabled	Controls the ability to use the USB port on the back of the Cisco IP Phones 8861 and 8865. Phones that do not support this feature do not display the field.
Side USB Port	Disabled Enabled	Enabled	Controls the ability to use the USB port on the side of the Cisco IP Phones 8851, 8851NR, 8861, 8865, and 8865NR. Phones that do not support this feature do not display the field.
Console Access	Disabled Enabled	Disabled	Specifies whether the serial console is enabled or disabled.
Bluetooth	Disabled Enabled	Enabled	Enables or disables the Bluetooth option on the phone. If disabled, the user cannot enable Bluetooth on the phone. Supported on the Cisco IP Phones 8845, 8851, 8861, and 8865. Phones that do not support this feature do not display the field.
Allow Bluetooth Contacts Import	Disabled Enabled	Enabled	Enables the user to import contacts from their connected mobile device using Bluetooth. When disabled, the user cannot import contacts from their connected mobile device on their phone. Supported on the Cisco IP Phones 8845, 8851, 8861, and 8865. Phones that do not support this feature do not display the field.
Allow Bluetooth Mobile Handsfree Mode	Disabled Enabled	Enabled	Enables users to take advantage of the acoustic properties of the phone with their mobile device or tablet. The user pairs the mobile device or tablet to the phone using Bluetooth. When disabled, the user cannot pair the mobile device or tablet with their phone. With a mobile device paired, the user can place and receive mobile calls on the phone. With a tablet, the user can route the audio from the tablet to the phone. Users can pair multiple mobile devices, tablets, and a Bluetooth headset to the phone. However, only one device and one headset can be connected at the same time. Phones that do not support this feature do not display the field.
Bluetooth Profiles	Handsfree Human Interface Device	Handsfree	Indicates which Bluetooth profiles on the phone are enabled or disabled. Phones that do not support this feature do not display the field.
Gratuitous ARP	Disabled Enabled	Disabled	Enables or disables the ability for the phone to learn MAC addresses from Gratuitous ARP. This capability is required to monitor or record voice streams.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Show All Calls on Primary Line	Disabled Enabled	Disabled	Specifies if all calls presented to this phone will be shown on the primary line or not. The purpose of this field is to make it easier for the end user to see all calls on all lines at a glance rather than having to choose a line to see the calls on that line. In other words, when multiple lines are configured on the phone, it typically makes more sense to be able to see all the calls on all lines in one combined display. When this feature is enabled, all calls will be shown on the primary line, but you can still choose a specific line to filter the display to show only the calls for that specific line.
HTTPS Server	HTTP and HTTPS enabled HTTPS only	HTTP and HTTPS enabled	Controls the type of communication to the phone. If you select HTTPS only, phone communication is more secure.
IPv6 Log Server	String of up to 256 characters		Identifies the IPv6 log server. The format for the address is: `[address]:<port>@@base=<0-7>;pfs=<0-1>`
Remote Log	Disabled Enabled	Disabled	Controls the ability to send logs to the syslog server.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Log Profile	Default Preset Telephony SIP UI Network Media Upgrade Accessory Security Wi-Fi VPN Energywise MobileRemoteAc	Preset	Specifies the predefined logging profile. • Default—Default debug logging level • Preset—Does not overwrite the phone local debug logging setting • Telephony—Logs information about Telephony or call features • SIP—Logs information about SIP signaling • UI—Logs information about the phone user interface • Network—Logs network information • Media—Logs media information • Upgrade—Logs upgrade information • Accessory—Logs accessory information • Security—Logs security information • Wi-Fi—Logs Wi-Fi information • VPN—Logs virtual private network information • Energywise—Logs energy-savings information • MobileRemoteAC—Logs Mobile and Remote Access through Expressway information
Advertise G.722 and iSAC Codecs	Use System Default Disabled Enabled	Use System Default	Indicates whether the phone advertises the G.722 and iSAC codecs to the Cisco Unified Communications Manager. • Use System Default—Defers to the setting specified in the enterprise parameter Advertise G.722 Codec. • Disabled—Does not advertise G.722 to the Cisco Unified Communications Manager. • Enabled—Advertises G.722 to the Cisco Unified Communications Manager. For more information, see the note that follows the table.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Detect Unified CM Connection Failure	Normal Delayed	Normal	Determines the sensitivity that the phone has for detecting a connection failure to Cisco Unified Communications Manager (Unified CM), which is the first step before device failover to a backup Unified CM/SRST occurs. • Normal—Detection of a Unified CM connection failure occurs at the standard system rate. Choose this value for faster recognition of a Unified CM connection failure. • Delayed—Detection of a Unified CM connection failover occurs approximately four times slower than Normal. Choose this value if you prefer failover to be delayed slightly to give the connection the opportunity to reestablish The precise time difference between Normal and Delayed connection failure detection depends on many variables that are constantly changing. This field only applies to the wired Ethernet connection.
Power Negotiation	Disabled Enabled	Enabled	Allows the phone to negotiate power using Link Level Endpoint Discovery Protocol (LLDP) and Cisco Discovery Protocol (CDP). Power Negotiation should not be disabled when the phone is connected to a switch that supports power negotiation. If disabled, the switch could shut off power to the phone.
Provide Dial Tone from Release Button	Disabled Enabled	Disabled	Controls whether the user hears dial tone when the Release key is pressed. • Disabled—User doesn't hear dial tone. • Enabled—User hears dial tone.
Background Image	String up to 64 characters		Specifies the default wallpaper file. When a default wallpaper is set, the user can't change the phone wallpaper.
Simplified New Call UI	Disabled Enabled	Disabled	Controls the user interface for off-hook dialing. When enabled, the user cannot select a number from the recent calls list. When enabled, this field provides a simplified window for the user to place a call. The user does not see the call history pop-up window that is displayed when the phone is taken off-hook. The pop-up window display is considered useful, so Simplified New Call UI is disabled by default.
Revert to All Calls	Disabled Enabled	Disabled	Specifies whether the phone will revert to All Calls after any call ends or not if the call is on a filter other than Primary line, All Calls, or Alerting Calls.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Show Call History for Selected Line Only	Disabled Enabled	Disabled	Controls the display of the Recents list. • Disabled—The Recents list shows the call history for all lines. • Enabled—The Recents list shows the call history for the selected line.
Actionable Incoming Call Alert	Disabled Show for all Incoming Call Show for Invisible Incoming Call	Show for all Incoming Call	Controls the type of incoming call alert that displays on the phone screen. The purpose of this field is to reduce the number of button presses that the end user requires to answer a call. • Disabled—The actionable incoming call alert is disabled and the user sees the traditional incoming call pop-up alert. • Show for all Incoming Call—The actionable incoming call alert displays for all calls regardless of visibility. • Show for Invisible Incoming Call—The actionable incoming call alert displays for calls not shown on the phone. This parameter behaves similarly to the incoming call alert pop-up notification.
DF bit	0 1	0	Controls how network packets are sent. Packets can be sent in chunks (fragments) of various sizes. When the DF bit is set to 1 in the packet header, the network payload does not fragment when going through network devices, such as switches and routers. Removing fragmenting avoids incorrect parsing on the receiving side, but results in slightly slower speeds. The DF bit setting does not apply to ICMP, VPN, VXC VPN, or DHCP traffic.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Default Line Filter	List of comma-separated phone device names		Indicates the list of phones that are in the default filter. When the default line filter is configured, users see a filter named `Daily schedule` in **Call notifications** in the **Settings** > **Preferences** menu of the phone. This daily schedule filter is in addition to the preset All Calls filter. If the default line filter is not configured, the phone checks all provisioned lines. If configured, the phone checks the lines set on Cisco Unified Communications Manager if the user selects Default filter as the active filter, or if there are no custom filters. Custom line filters enable you to filter on high-priority lines to reduce alert activity. You can set the alerting call notification priority on a subset of lines covered by an alert filter. The custom filter generates either traditional pop-up alerts or actionable alerts for incoming calls on the selected lines. For each filter, only the covered subset of lines will generate an alert. This feature provides a way for users with multiple lines to reduce alert activity by filtering and displaying alerts only from high-priority lines. The end users can configure this themselves. Alternatively, you can program the default line filter and push the filter down to the phone.
Lowest Alerting Line State Priority	Disabled Enabled	Disabled	Specifies the alert state when using shared lines. • Disabled—When there is an incoming call alerting on the shared line, the LED/Line state icon reflects the alerting state instead of Remote-In-Use. • Enabled—When there is an incoming call alerting on the shared line, the user sees the Remote-In-Use icon.
One Column Display for KEM	Disabled Enabled	Disabled	Controls the display on the Key Expansion Module. • Disabled—The expansion module uses two-column mode. • Enabled—The expansion module uses one-column mode. Phones that do not support this feature do not display the field.
Energy Efficient Ethernet (EEE): PC Port	Disabled Enabled	Disabled	Controls EEE on the PC port.
Energy Efficient Ethernet (EEE): SW Port	Disabled Enabled	Disabled	Controls EEE on the switch port.
Start Video Port			Defines the start of the port range for video calls. Phones that do not support this feature do not display the field.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Stop Video Port			Defines the end of the port range for video calls. Phones that do not support this feature do not display the field.
User Credentials Persistent for Expressway Sign in	Disabled Enabled	Disabled	Controls if the phone stores the users' sign-in credentials. When disabled, the user is always sees the prompt to sign into the Expressway server for Mobile and Remote Access (MRA). If you would like to make it easier for users to log in, you enable this field so that the Expressway login credentials are persistent. The user then only has to enter their login credentials the first time. Any time after that (when the phone is powered on off-premise), the login information is prepopulated on the Sign-in screen. For more information, see the Mobile and Remote Access Through Expressway, on page 167.
Customer support upload URL	String, up to 256 characters		Provides the URL for the Problem Report Tool (PRT). If you deploy devices with Mobile and Remote Access through Expressway, you must also add the PRT server address to the HTTP Server Allow list on the Expressway server. For more information, see the Mobile and Remote Access Through Expressway, on page 167.
Web Admin	Disabled Enabled	Disabled	Enables or disables administrator access to the phone web pages through a web browser For more information, see the Configure the Administration Page for Phone, on page 102. Phones that do not support this feature do not display the field.
Admin Password	String of 8–127 characters		Defines the administrator password when you access the phone web pages as an administrator. Phones that do not support this feature do not display the field.
WLAN SCEP Server	String of up to 256 characters		Specifies the SCEP Server that the phone uses to obtain certificates for WLAN authentication. Enter the hostname or the IP address (using standard IP addressing format) of the server. Phones that do not support this feature do not display the field.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
WLAN Root CA Fingerprint (SHA256 or SHA1)	String of up to 95 characters		Specifies the SHA256 or SHA1 fingerprint of the Root CA to use for validation during the SCEP process when issuing certificates for WLAN authentication. We recommend that you use the SHA256 fingerprint, which can be obtained via OpenSSL (e.g. openssl x509 -in rootca.cer -noout -sha256 -fingerprint) or using a Web Browser to inspect the certificate details. Enter the 64 hexadecimal character value for the SHA256 fingerprint or the 40 hexadecimal character value for the SHA1 fingerprint with a common separator (colon, dash, period, space) or without a separator. If using a separator, then the separator should be consistently placed after every 2, 4, 8, 16, or 32 hexadecimal characters for a SHA256 fingerprint or every 2, 4, or 8 hexadecimal characters for a SHA1 fingerprint. Phones that do not support this feature do not display the field.
WLAN Authentication Attempts			Phones that do not support this feature do not display the field.
WLAN Profile 1 Prompt Mode	Disabled Enabled	Disabled	Phones that do not support this feature do not display the field.
Line Mode	Session Line Mode Enhanced Line Mode	Session Line Mode	Controls the line display on the phone. • Session Line Mode—The buttons on one side of the screen are line keys. • Enhanced Line Mode—The buttons on both sides of the phone screen are line keys. Predictive dialing and Actionable incoming call alerts are enabled by default in Enhanced line mode.
Admin Configurable Ringer	Disabled Sunrise Chirp1 Chirp2	Disabled	Controls the ringtone and the ability for users to set the ringtone. • When set to **Disabled**, users can configure the default ringtone on their phones. • For all other values, users cannot change the ringtone. The **Ringtone** menu item in the **Settings** menu is grayed out.
Customer Support Use	String of up to 64 characters	Empty	For Cisco TAC use only.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Disable TLS Ciphers	See Disable Transport Layer Security Ciphers, on page 156.	None	Disables the selected TLS cipher. Disable more than one cipher suite by selecting and holding the **Ctrl** key on your computer keyboard. If you select all of the phone ciphers, then phone TLS service is impacted.
Lower Your Voice Alert	Enabled Disabled	Enabled	Controls the Lower your voice feature. • Disabled: • The phone doesn't display the **Lower your voice** menu item in the **Settings** menu. • Users won't see the message on their screen when they speak loudly. • Enabled: • Users control the feature from the **Lower your voice** menu item in the **Settings** menu. By default, the field is set to **On**.
Mark Call As Spam	Enabled Disabled	Enabled	Controls the Mark call as spam feature. • Disabled: • The phone doesn't display the **Mark spam** softkey. • The **Spam list** item in the **Settings** menu doesn't display. • If there was a spam list, the list is cleared and can't be recovered. • Enabled: • The phone displays the **Mark spam** softkey. • The **Spam list** item in the **Settings** menu displays.
Dedicate one line for Call Park	Disabled Enabled	Enabled	Controls whether a parked call occupies one line or not. For more information, see the Cisco Unified Communications Manager documentation.

Field Name	Field Type or Choices	Default	Description and Usage Guidelines
Line Text Label Display in ELM	Disabled Enabled	Enabled	Controls the line label display during a call when Enhanced Line Mode is configured • Enabled • If the caller's name is configured, it displays the name in the first line of the call session and the local line label in the second line. • If the caller's name is not configured, it displays the remote number in the first line and the local line label in the second line. • Disabled • If the caller's name is configured, it displays the name in the first line of the call session and number in the second line. • If the caller's name is not configured, it displays **only** the remote number. This field is required.



Note Codec negotiation involves two steps:

1. The phone advertises the supported codec to the Cisco Unified Communications Manager. Not all endpoints support the same set of codecs.
2. When the Cisco Unified Communications Manager gets the list of supported codecs from all phones involved in the call attempt, it chooses a commonly supported codec based on various factors, including the region pair setting.

Feature Configuration Best Practices

You can set up the phone features to suit your users' needs. But we have some recommendations for certain situations and deployments that might help you.

High Call Volume Environments

In a high call volume environment, we recommend that you set up some features in a specific way.

Field	Administration Area	Recommended Setting
Always Use Prime Line	Device Information	Off or On For more information, see Field: Always Use Prime Line, on page 156.
Actionable Incoming Call Alert	Product Specific Configuration Layout	Show for all Incoming Call
Show All Calls on Primary Line	Product Specific Configuration Layout	Enabled
Revert to All Calls	Product Specific Configuration Layout	Enabled

Multiline Environments

In a multiline environment, we recommend that you set up some features in a specific way.

Field	Administration Area	Recommended Setting
Always Use Prime Line	Device Information	Off For more information, see Field: Always Use Prime Line, on page 156.
Actionable Incoming Call Alert	Product Specific Configuration Layout	Show for all Incoming Call
Show All Calls on Primary Line	Product Specific Configuration Layout	Enabled
Revert to All Calls	Product Specific Configuration Layout	Enabled

Session Line Mode Environment

Enhanced Line Mode is the preferred tool for handling most call environments. However, if Enhanced Line Mode does not suit your needs, then you can use Session line mode.

Field	Administration Area	Recommended Setting for Session Line Mode
Show All Calls on Primary Line	Product Specific Configuration Layout	Disabled
Revert to All Calls	Product Specific Configuration Layout	Disabled
Actionable Incoming Call Alert	Product Specific Configuration Layout	Enabled by default (Firmware Release 11.5(1) and later).

Related Topics

Set Up Additional Line Keys, on page 189
Features Available in Enhanced Line Mode, on page 190

Field: Always Use Prime Line

This field specifies whether the primary line on an IP phone is chosen when a user goes off-hook. If this parameter is set to True, when a phone goes off-hook, the primary line is chosen and becomes the active line. Even if a call rings on the second line of the user, when the phone goes off-hook, it makes only the first line active. It does not answer the inbound call on the second line. In this case, the user must choose the second line to answer the call. The default value is set to False.

The purpose of the Always Use Prime Line field is very similar to the combination of Show All Calls on the Primary Line and Revert to All Calls when both of those two features are enabled. However, the main difference is that when Always Use Prime Line is enabled, inbound calls are not answered on the second line. Only dial tone is heard on the prime line. There are certain high call volume environments where this is the desired user experience. In general, it is best to leave this field disabled except for high call volume environments that require this feature.

Disable Transport Layer Security Ciphers

You can disable Transport Layer Security (TLS) ciphers with the **Disable TLS Ciphers** parameter. This allows you to tailor your security for known vulnerabilities, and to align your network with your company's policies for ciphers.

None is the default setting.

Disable more than one cipher suite by selecting and holding the **Ctrl** key on your computer keyboard. If you select all of the phone ciphers, then phone TLS service is impacted. Your choices are:

- None
- TLS_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_RSA_WITH_AES_128_CBC_SHA
- TLS_RSA_WITH_AES_256_CBC_SHA
- TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384
- TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384

For more information about phone security, see *Cisco IP Phone 7800 and 8800 Series Security Overview White Paper* (https://www.cisco.com/c/en/us/products/collaboration-endpoints/unified-ip-phone-8800-series/white-paper-listing.html).

Enable Call History for Shared Line

Allows you to view your shared line activity in the Call History. This feature:

- Logs missed calls for a shared line.

- Logs all answered and placed calls for a shared line.

Before you begin

Disable Privacy before you enable Call History for Shared Line. Otherwise Call History doesn't display the calls other users answer.

Procedure

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

Step 2 Locate the phone to be configured.

Step 3 Navigate to the Record Call Log from Shared Line drop-down in the Product Specific Configuration area.

Step 4 Select **Enabled** from the drop-down list.

Step 5 Select **Save**.

Schedule Power Save for Cisco IP Phone

To conserve power and ensure the longevity of the phone screen display, you can set the display to turn off when it is not needed.

You can configure settings in Cisco Unified Communications Manager Administration to turn off the display at a designated time on some days and all day on other days. For example, you may choose to turn off the display after business hours on weekdays and all day on Saturdays and Sundays.

You can take any of these actions to turn on the display any time it is off:

- Press any button on the phone.

  The phone takes the action designated by that button in addition to turning on the display.

- Lift the handset.

When you turn the display on, it remains on until the phone has remained idle for a designated length of time, then it turns off automatically.

For more information, see Product Specific Configuration, on page 136

Procedure

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

Step 2 Locate the phone that you need to set up.

Step 3 Navigate to the Product Specific Configuration area and set the following fields:

- Days Display Not Active
- Display On Time
- Display On Duration

- Display Idle Timeout

Table 32: PowerSave Configuration Fields

Field	Description
Days Display Not Active	Days that the display does not turn on automatically at the time specified in the Display On Time field. Choose the day or days from the drop-down list. To choose more than one day, Ctrl-click each day that you want.
Display On Time	Time each day that the display turns on automatically (except on the days specified in the Days Display Not Active field). Enter the time in this field in 24-hour format, where 0:00 is midnight. For example, to automatically turn the display on at 07:00a.m., (0700), enter **07:00**. To turn the display on at 02:00p.m. (1400), enter **14:00**. If this field is blank, the display will automatically turn on at 0:00.
Display On Duration	Length of time that the display remains on after turning on at the time specified in the Display On Time field. Enter the value in this field in the format *hours:minutes*. For example, to keep the display on for 4 hours and 30 minutes after it turns on automatically, enter **04:30**. If this field is blank, the phone will turn off at the end of the day (0:00). **Note** If Display On Time is 0:00 and the display on duration is blank (or 24:00), the display will remain on continuously.
Display Idle Timeout	Length of time that the phone is idle before the display turns off. Applies only when the display was off as scheduled and was turned on by a user (by pressing a button on the phone or lifting the handset). Enter the value in this field in the format *hours:minutes*. For example, to turn the display off when the phone is idle for 1 hour and 30 minutes after a user turns the display on, enter **01:30**. The default value is 01:00.

Step 4 Select **Save**.

Step 5 Select **Apply Config**.

Step 6 Restart the phone.

Schedule EnergyWise on Cisco IP Phone

To reduce power consumption, configure the phone to sleep (power down) and wake (power up) if your system includes an EnergyWise controller.

You configure settings in Cisco Unified Communications Manager Administration to enable EnergyWise and configure sleep and wake times. These parameters are closely tied to the phone display configuration parameters.

When EnergyWise is enabled and a sleep time is set, the phone sends a request to the switch to wake it up at the configured time. The switch returns either an acceptance or a rejection of the request. If the switch rejects the request or if the switch does not reply, the phone does not power down. If the switch accepts the request, the idle phone goes to sleep, thus reducing the power consumption to a predetermined level. A phone that is not idle sets an idle timer and goes to sleep after the idle timer expires.

To wake up the phone, press Select. At the scheduled wake time, the system restores power to the phone, waking it up.

For more information, see Product Specific Configuration, on page 136

Procedure

Step 1 From the Cisco Unified Communications Manager Administration, select **Device** > **Phone**.

Step 2 Locate the phone that you need to set up.

Step 3 Navigate to the Product Specific Configuration area and set the following fields.

- Enable Power Save Plus
- Phone On Time
- Phone Off Time
- Phone Off Idle Timeout
- Enable Audible Alert
- EnergyWise Domain
- EnergyWise Secret
- Allow EnergyWise Overrides

Table 33: EnergyWise Configuration Fields

Field	Description
Enable Power Save Plus	Selects the schedule of days for which the phone powers off. Select multiple days by pressing and holding the Control key while clicking on the days for the schedule. By default, no days are selected. When Enable Power Save Plus is checked, you receive a message that warns about emergency (e911) concerns. **Caution** While Power Save Plus Mode (the "Mode") is in effect, endpoints that are configured for the mode are disabled for emergency calling and from receiving inbound calls. By selecting this mode, you agree to the following: (i) You take full responsibility for providing alternate methods for emergency calling and receiving calls while the mode is in effect; (ii) Cisco has no liability in connection with your selection of the mode and all liability in connection with enabling the mode is your responsibility; and (iii) You fully inform users of the effects of the mode on calls, calling and otherwise. **Note** To disable Power Save Plus, you must uncheck the Allow EnergyWise Overrides check box. If the Allow EnergyWise Overrides remains checked but no days are selected in the Enable Power Save Plus field, Power Save Plus is not disabled.
Phone On Time	Determines when the phone automatically turns on for the days that are in the Enable Power Save Plus field. Enter the time in this field in 24-hour format, where 00:00 is midnight. For example, to automatically power up the phone at 07:00 a.m. (0700), enter 07:00. To power up the phone at 02:00 p.m. (1400), enter 14:00. The default value is blank, which means 00:00. **Note** The Phone On Time must be at least 20 minutes later than the Phone Off Time. For example, if the Phone Off Time is 07:00, the Phone On Time must be no earlier than 07:20.
Phone Off Time	The time of day that the phone powers down for the days that are selected in the Enable Power Save Plus field. If the Phone On Time and the Phone Off Time fields contain the same value, the phone does not power down. Enter the time in this field in 24-hour format, where 00:00 is midnight. For example, to automatically power down the phone at 7:00 a.m. (0700), enter 7:00. To power down the phone at 2:00 p.m. (1400), enter 14:00. The default value is blank, which means 00:00. **Note** The Phone On Time must be at least 20 minutes later than the Phone Off Time. For example, if the Phone Off Time is 7:00, the Phone On Time must be no earlier than 7:20.

Field	Description
Phone Off Idle Timeout	The length of time that the phone must be idle before the phone powers down. The timeout occurs under the following conditions: • When the phone was in Power Save Plus mode, as scheduled, and was taken out of Power Save Plus mode because the phone user pressed the **Select** key. • When the phone is repowered by the attached switch. • When the Phone Off Time is reached but the phone is in use. The range of the field is 20 to 1440 minutes. The default value is 60 minutes.
Enable Audible Alert	When enabled, instructs the phone to play an audible alert starting 10 minutes before the time that the Phone Off Time field specifies. The audible alert uses the phone ringtone, which briefly plays at specific times during the 10-minute alerting period. The alerting ringtone plays at the user-designated volume level. The audible alert schedule is: • At 10 minutes before power down, play the ringtone four times. • At 7 minutes before power down, play the ringtone four times. • At 4 minutes before power down, play the ringtone four times. • At 30 seconds before power down, play the ringtone 15 times or until the phone powers off. This check box applies only if the Enable Power Save Plus list box has one or more days selected.
EnergyWise Domain	The EnergyWise domain that the phone is in. The maximum length of this field is 127 characters.
EnergyWise Secret	The security secret password that is used to communicate with the endpoints in the EnergyWise domain. The maximum length of this field is 127 characters.

Field	Description
Allow EnergyWise Overrides	This check box determines whether you allow the EnergyWise domain controller policy to send power level updates to the phones. The following conditions apply: • One or more days must be selected in the Enable Power Save Plus field. • The settings in Cisco Unified Communications Manager Administration take effect on schedule even if EnergyWise sends an override. For example, assuming the Phone Off Time is set to 22:00 (10:00 p.m.), the value in the Phone On Time field is 06:00 (6:00 a.m.), and the Enable Power Save Plus has one or more days selected. • If EnergyWise directs the phone to turn off at 20:00 (8:00 p.m.), that directive remains in effect (assuming no phone user intervention occurs) until the configured Phone On Time at 6:00 a.m. • At 6:00 a.m., the phone turns on and resumes receiving the power level changes from the settings in Unified Communications Manager Administration. • To change the power level on the phone again, EnergyWise must reissue a new power level change command. **Note** To disable Power Save Plus, you must uncheck the Allow EnergyWise Overrides check box. If the Allow EnergyWise Overrides remains checked but no days are selected in the Enable Power Save Plus field, Power Save Plus is not disabled.

Step 4 Select **Save**.

Step 5 Select **Apply Config**.

Step 6 Restart the phone.

Set Up Do Not Disturb

When Do Not Disturb (DND) is turned on, either no audible rings occur during the ringing-in state of a call, or no audible or visual notifications of any type occur.

When Do Not Disturb (DND) is enabled, the header section of the phone screen changes color, and `Do not disturb` is displayed on the phone.

You can configure the phone with a phone-button template with DND as one of the selected features.

For more information, see the Do Not Disturb information in the documentation for your particular Cisco Unified Communications Manager release.

Procedure

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

Step 2 Locate the phone to be configured.

Step 3 Set the following parameters.

- Do Not Disturb: This check box allows you to enable DND on the phone.
- DND Option: Ring Off, Call Reject, or Use Common Phone Profile Setting.

  Do not choose Call Reject if you want priority (MLPP) calls to ring this phone when DND is turned on.
- DND Incoming Call Alert: Choose the type of alert, if any, to play on a phone for incoming calls when DND is active.

  Note This parameter is located on in the Common Phone Profile window and the Phone Configuration window. The Phone Configuration window value takes precedence.

Step 4 Select **Save**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Enable Agent Greeting

The Agent Greeting feature allows an agent to create and update a prerecorded greeting that plays at the beginning of a call, such as a customer call, before the agent begins the conversation with the caller. The agent can prerecord a single greeting or multiple greetings, as needed, and create and update the greetings.

When a customer calls, the agent and the caller hear the prerecorded greeting. The agent can remain on mute until the greeting ends or the agent can answer the call over the greeting.

All codecs supported for the phone are supported for Agent Greeting calls.

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

Procedure

Step 1 From Cisco Unified Communications Manager Administration, select **Device** > **Phone**.

Step 2 Locate the IP phone that you want to configure.

Step 3 Scroll to the Device Information Layout pane and set **Built In Bridge** to On or Default.

Step 4 Select **Save**.

Step 5 Check the setting of the bridge:

a) Choose **System** > **Service Parameters**.
b) Select the appropriate Server and Service.
c) Scroll to the Clusterwide Parameters (Device - Phone) pane and set **Builtin Bridge Enable** to On.
d) Select **Save**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Set Up Monitoring and Recording

The Monitoring and Recording feature allows a supervisor to monitor an active call silently. Neither party on the call can hear the supervisor. The user may receive an audible alert during a call when it is being monitored.

When a call is secure, a lock icon displays. Callers may also receive an audible alert to indicate that the call is being monitored. The connected parties may also receive an audible alert that indicates that the call is secure and is being monitored.

When an active call is being monitored or recorded, the user can receive or place intercom calls; however, if the user places an intercom call, the active call is put on hold. This action causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the person being monitored must resume the call.

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

The following procedure adds a user to the standard monitoring user groups.

Before you begin

The Cisco Unified Communications Manager must be configured to support Monitoring and Recording.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **User Management** > **Application User**.

Step 2 Check the Standard CTI Allow Call Monitoring user group and the Standard CTI Allow Call Recording user groups.

Step 3 Click **Add Selected**.

Step 4 Click **Add to User Group**.

Step 5 Add the user phones to the list of Application Users controlled devices.

Step 6 Select **Save**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Set Up Call Forward Notification

You can control the call forward settings.

Procedure

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

Step 2 Locate the phone to be set up.

Step 3 Configure the Call Forward Notification fields.

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

Step 4 Select **Save**.

Enable BLF for Call Lists

The BLF for Call Lists field also controls the Line Status for Corporate Directory feature.

Procedure

Step 1 In the Cisco Unified Communications Manager Administration, select **System** > **Enterprise Parameters**.

Step 2 For the BLF for Call Lists field, enable or disable the feature.

By default, the feature is disabled.

Parameters that you set in the Product Specific Configuration area may also appear in the Device Configuration window for various devices and in the Enterprise Phone Configuration window. If you set these same parameters in these other windows as well, the setting that takes precedence is determined in the following order:

a. Device Configuration window settings

b. Common Phone Profile window settings

c. Enterprise Phone Configuration window settings

Step 3 Select **Save**.

Set Up Energy Efficient Ethernet for Switch and PC Port

IEEE 802.3az Energy Efficient Ethernet (EEE) is an extension of the IEEE 802.3 standard that provides a method for reducing energy usage without reducing the vital function of network interfaces. Configurable EEE enables the administrator to control EEE functions on personal computer port and switch port.

Note Administrators must confirm that the Override Check box is checked on all applicable UCM pages or EEE will not function.

The administrator controls the EEE functions with the following two parameters:

- **Energy Efficient Ethernet**: PC Port: Provides seamless connection with personal computers. Administrator can select Enabled or Disabled options to control the function.
- **Energy Efficient Ethernet**: Switch Port: Provides seamless connection

For more information, seeProduct Specific Configuration, on page 136

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select one of the following windows:

- **Device** > **Phone**
- **Device** > **Device Settings** > **Common Phone Profile**
- **System** > **Enterprise Phone Configurations**

If you configure the parameter in multiple windows, the precedence order is:

a. **Device** > **Phone**

b. **Device** > **Device Settings** > **Common Phone Profile**

c. **System** > **Enterprise Phone Configurations**

Step 2 If required, locate the phone.

Step 3 Set the **Energy Efficient Ethernet: PC Port** and **Energy Efficient Ethernet: Switch Port** fields.

- Energy Efficient Ethernet: PC Port
- Energy Efficient Ethernet: Switch Port

Step 4 Select **Save**.

Step 5 Select **Apply Config**.

Step 6 Restart the phone.

Set Up RTP/sRTP Port Range

You configure the Real-Time Transport Protocol (RTP) and secure Real-Time Transport Protocol (sRTP) port values in the SIP profile. RTP and sRTP port values range from 2048 to 65535, with a default range of 16384 to 32764. Some port values within the RTP and sRTP port range are designated for other phone services. You cannot configure these ports for RTP and sRTP.

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

Procedure

Step 1 Select **Device** > **Device Settings** > **SIP Profile**

Step 2 Choose the search criteria to use and click **Find**.

Step 3 Select the profile to modify.

Step 4 Set the Start Media Port and Stop Media Port to contain the start and end of the port range.

The following list identifies the UDP ports that are used for other phone services and thus not available for RTP and sRTP use:

port 4051

used for the Peer Firmware Sharing (PFS) feature

port 5060

used for SIP over UDP transport

port range 49152 to 53247

used for local ephemeral ports

port range 53248 to 65535

used for the VxC single tunnel VPN feature

Step 5 Click **Save**.

Step 6 Click **Apply Config**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Mobile and Remote Access Through Expressway

Mobile and Remote Access Through Expressway(MRA) lets remote workers easily and securely connect into the corporate network without using a virtual private network (VPN) client tunnel. Expressway uses Transport Layer Security (TLS) to secure network traffic. For a phone to authenticate an Expressway certificate and establish a TLS session, a public Certificate Authority that the phone firmware trusts must sign the Expressway certificate. It is not possible to install or trust other CA certificates on phones for authenticating an Expressway certificate.

The list of CA certificates embedded in the phone firmware is available at http://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8800-series/products-technical-reference-list.html.

Mobile and Remote Access Through Expressway (MRA)works with Cisco Expressway. You must be familiar with the Cisco Expressway documentation, including the *Cisco Expressway Administrator Guide* and the *Cisco Expressway Basic Configuration Deployment Guide*. Cisco Expressway documentation is available at http://www.cisco.com/c/en/us/support/unified-communications/expressway-series/tsd-products-support-series-home.html.

Only the IPv4 protocol is supported for Mobile and Remote Access Through Expressway users.

For additional information about working with Mobile and Remote Access Through Expressway, see:

- *Cisco Preferred Architecture for Enterprise Collaboration, Design Overview*
- *Cisco Preferred Architecture for Enterprise Collaboration, CVD*
- *Unified Communications Mobile and Remote Access via Cisco VCS Deployment Guide*
- *Cisco TelePresence Video Communication Server (VCS), Configuration Guides*
- *Mobile and Remote Access Through Cisco Expressway Deployment Guide*

During the phone registration process, the phone synchronizes the displayed date and time with the Network Time Protocol (NTP) server. With MRA, the DHCP option 42 tag is used to locate the IP addresses of the NTP servers designated for time and date synchronization. If the DHCP option 42 tag is not found in the configuration information, the phone looks for the 0.tandberg.pool.ntp.org tag to identify the NTP servers.

After registration, the phone uses information from the SIP message to synchronize the displayed date and time unless an NTP server is configured in the Cisco Unified Communications Manager phone configuration.

Note If the phone security profile for any of your phones has TFTP Encrypted Config checked, you cannot use the phone with Mobile and Remote Access. The MRA solution does not support device interaction with Certificate Authority Proxy Function (CAPF).

Mobile and Remote Access Through Expressway supports Enhanced line mode.

SIP OAuth mode is supported for MRA. This mode allows you to use OAuth access tokens for authentication in secure environments.

Note For SIP OAuth in Mobile and Remote Access (MRA) mode, use only Activation Code Onboarding with Mobile and Remote Access when you deploy the phone. Activation with a username and password is not supported.

SIP OAuth mode requires Expressway x14.0(1) and later, or Cisco Unified Communications Manager 14.0(1) and later.

For additional information on SIP OAuth mode see *Feature Configuration Guide for Cisco Unified Communications Manager*, Release 14.0(1) or later.

Deployment Scenarios

The following sections show various deployment scenarios for Mobile and Remote Access Through Expressway.

On-Premises User Logs In to the Enterprise Network

After Mobile and Remote Access Through Expressway is deployed, log in to the enterprise network when on-premises. The phone detects the network, and registers with Cisco Unified Communications Manager.

Off-Premises User Logs In to the Enterprise Network

When you are away from the office, the phone detects that it is in off-premises mode. The Mobile and Remote Access Through Expressway Sign-In window appears, and you connect to the corporate network.

Note the following:

- You must have a valid service domain, username, and password to connect to the network.
- Reset the service mode to clear the Alternate TFTP setting before you try to access the company network. This clears the Alternate TFTP Server setting so the phone detects the off-premises network, and it stops the phone from making a VPN connection. Skip this step if a phone is being deployed for the first time.
- If you have DHCP option 150 or option 66 enabled on your network router, you may not be able to sign into the corporate network. Reset your service mode to enter MRA mode.

Off-Premises User Logs In to the Enterprise Network with VPN

When you are off-premises, log in to the enterprise network with VPN, after deploying Mobile and Remote Access Through Expressway.

Perform a Basic Reset to reset your phone configurations if your phone experiences an error.

You must configure the Alternate TFTP setting (**Admin settings** > **Network settings** > **IPv4**, field **Alternate TFTP server 1**).

Related Topics

Basic Reset, on page 259

Media Paths and Interactive Connectivity Establishment

You can deploy Interactive Connectivity Establishment (ICE) to improve the reliability of Mobile and Remote Access (MRA) calls that cross a firewall or Network Address Translation (NAT). ICE is an optional deployment that uses Serial Tunneling and Traversal Using Relays around NAT services to select the best media path for a call.

Secondary Turn Server and Turn Server Failover is not supported.

For more information about MRA and ICE, see *System Configuration Guide for Cisco Unified Communications Manager, Release 12.0(1)* or later. You can also find additional information in the Internet Engineering Task Force (IETF) Request for Comment documents:

- *Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)*(RFC 5766)
- *Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols* (RFC 5245)

Phone Features Available for Mobile and Remote Access Through Expressway

Mobile and Remote Access Through Expressway provides secure VPN-less access to collaboration services for Cisco mobile and remote users. But to preserve network security, it limits access to some phone features.

The following list shows the phone features available with Mobile and Remote Access Through Expressway.

Table 34: Feature Support and Mobile and Remote Access Through Expressway

Phone Feature	Phone Firmware Release
Abbreviated Dialing	10.3(1) and later
Answer Oldest	11.5(1)SR1 and later
Assisted Directed Call Park	10.3(1) and later
Auto Answer	11.5(1)SR1 and later
Barge and cBarge	11.5(1)SR1 and later
Busy Lamp Field (BLF)	10.3(1) and later
Busy Lamp Field (BLF) Pickup	10.3(1) and later
Busy Lamp Field (BLF) Speed Dial	10.3(1) and later
Call Back	10.3(1) and later
Call Forward	10.3(1) and later
Call Forward Notification	10.3(1) and later
Call Park	10.3(1) and later
Call Pickup	10.3(1) and later
Cisco Unified Serviceability	11.5(1)SR1 and later
Client Access License (CAL)	11.5(1)SR1 and later
Conference	10.3(1) and later
Conference List / Remove Participant	11.5(1)SR1 and later
Corporate Directory	11.5(1)SR1 and later
CTI Applications (CTI Controlled)	11.5(1)SR1 and later
Direct Transfer	10.3(1) and later
Directed Call Park	10.3(1) and later
Distinctive Ring	11.5(1)SR1 and later
Divert	10.3(1) and later
Enhanced line mode	12.1(1) and later
Divert	10.3(1) and later
Forced Access Codes and Client Matter Codes	11.5(1)SR1 and later

Phone Feature	Phone Firmware Release
Group Call Pickup	10.3(1) and later
Hold/Resume	10.3(1) and later
Hold Reversion	10.3(1) and later
Immediate Divert	10.3(1) and later
Join	10.3(1) and later
Malicious Caller Identification (MCID)	11.5(1)SR1 and later
Meet Me Conference	10.3(1) and later
Message Waiting Indicator	10.3(1) and later
Mobile Connect	10.3(1) and later
Mobile Voice Access	10.3(1) and later
Multilevel Precedence and Preemption (MLPP)	11.5(1)SR1 and later
Multiline	11.5(1)SR1 and later
Music On Hold	10.3(1) and later
Mute	10.3(1) and later
Network profiles (Automatic)	11.5(1)SR1 and later
Off-hook Dialing	10.3(1) and later
On-hook Dialing	10.3(1) and later
Plus Dialing	10.3(1) and later
Privacy	11.5(1)SR1 and later
Private Line Automated Ringdown (PLAR)	11.5(1)SR1 and later
Redial	10.3(1) and later
Speed Dial (does not support a pause)	10.3(1) and later
Services URL button	11.5(1)SR1 and later
Transfer	10.3(1) and later
Uniform Resource Identifier (URI) Dialing	10.3(1) and later

Configure User Credentials Persistent for Expressway Sign-In

When a user signs in to the network with Mobile and Remote Access Through Expressway, the user is prompted for a service domain, username, and password. If you enable the User Credentials Persistent for Expressway

Sign-In parameter, user login credentials are stored so that they do not need to reenter this information. This parameter is disabled by default.

You can set up credentials to persist for a single phone, a group of phones, or all phones.

Related Topics

Phone Feature Configuration, on page 135
Product Specific Configuration, on page 136

Generate a QR Code for MRA Sign-In

Users who have a phone with a camera can scan a QR code to sign into MRA, instead of entering the service domain and their username manually.

Procedure

Step 1 Use a QR code generator to generate a QR code with either the service domain, or the service domain and username separated by a comma. For example: mra.example.com or mra.example.com,username.

Step 2 Print the QR code and provide it to the user.

Problem Report Tool

Users submit problem reports to you with the Problem Report Tool.

Note The Problem Report Tool logs are required by Cisco TAC when troubleshooting problems. The logs are cleared if you restart the phone. Collect the logs before you restart the phones.

To issue a problem report, users access the Problem Report Tool and provide the date and time that the problem occurred, and a description of the problem.

If the PRT upload fails, you can access the PRT file for the phone from the URL **`http://<phone-ip-address>/FS/<prt-file-name>`**. This URL is displayed on the phone in the following cases:

- If the phone is in the factory default state. The URL is active for 1 hour. After 1 hour, the user should try to submit the phone logs again.
- If the phone has downloaded a configuration file and the call control system allows web access to the phone.

You must add a server address to the **Customer Support Upload URL** field on Cisco Unified Communications Manager.

If you are deploying devices with Mobile and Remote Access through Expressway, you must also add the PRT server address to the HTTP Server Allow list on the Expressway server.

Configure a Customer Support Upload URL

You must use a server with an upload script to receive PRT files. The PRT uses an HTTP POST mechanism, with the following parameters included in the upload (utilizing multipart MIME encoding):

- devicename (example: “SEP001122334455”)
- serialno (example: “FCH12345ABC”)
- username (the username configured in Cisco Unified Communications Manager, the device owner)
- prt_file (example: “probrep-20141021-162840.tar.gz”)

A sample script is shown below. This script is provided for reference only. Cisco does not provide support for the upload script installed on a customer's server.

```
<?php

// NOTE: you may need to edit your php.ini file to allow larger
// size file uploads to work.
// Modify the setting for upload_max_filesize
// I used:  upload_max_filesize = 20M

// Retrieve the name of the uploaded file
$filename = basename($_FILES['prt_file']['name']);

// Get rid of quotes around the device name, serial number and username if they exist
$devicename = $_POST['devicename'];
$devicename = trim($devicename, "'\"");

$serialno = $_POST['serialno'];
$serialno = trim($serialno, "'\"");

$username = $_POST['username'];
$username = trim($username, "'\"");

// where to put the file
$fullfilename = "/var/prtuploads/".$filename;

// If the file upload is unsuccessful, return a 500 error and
// inform the user to try again

if(!move_uploaded_file($_FILES['prt_file']['tmp_name'], $fullfilename)) {
        header("HTTP/1.0 500 Internal Server Error");
        die("Error: You must select a file to upload.");
}

?>
```



Note The phones only support HTTP URLs.

Procedure

Step 1 Set up a server that can run your PRT upload script.

Step 2 Write a script that can handle the parameters listed above, or edit the provided sample script to suit your needs.

Step 3 Upload your script to your server.

Step 4 In Cisco Unified Communications Manager, go to the Product Specific Configuration Layout area of the individual device configuration window, Common Phone Profile window, or Enterprise Phone Configuration window.

Step 5 Check **Customer support upload URL** and enter your upload server URL.

Example:

http://example.com/prtscript.php

Step 6 Save your changes.

Set the Label for a Line

You can set up a phone to display a text label instead of the directory number. Use this label to identify the line by name or function. For example, if your user shares lines on the phone, you could identify the line with the name of the person that shares the line.

When adding a label to a key expansion module, only the first 25 characters are displayed on a line.

Procedure

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

Step 2 Locate the phone to be configured.

Step 3 Locate the line instance and set the Line Text Label field.

Step 4 (Optional) If the label needs to be applied to other devices that share the line, check the Update Shared Device Settings check box and click **Propagate Selected**.

Step 5 Select **Save**.

Set Up Dual Bank Information

To set up Dual Bank Information, follow these steps:

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **Device** > **Device Defaults**.

Step 2 Check the load information in the Inactive Load Information field.

Step 3 Choose **Bulk Administration** > **Import/Export** > **Export** > **Device Defaults**, and schedule an export job.

Step 4 Download the exported tar file and untar it.

Step 5 Check the file format in the exported CSV file and verify that the CSV file has an Inactive Load Information column with the correct value.

Note The CSV file value must match the Device Default value in the Cisco Unified Communications Manager Administration window.

Park Monitoring

Park monitoring is supported only when a Cisco IP phone parks a call. Park monitoring then monitors the status of a parked call. The park monitoring call bubble does not clear until the parked call gets retrieved or is abandoned by the parked call. This parked call can be retrieved by using the same call bubble on the phone that parked the call.

Set Up Park Monitoring Timers

Cisco Unified Communications Manager Administration provides three cluster-wide service timer parameters for park monitoring: Park Monitoring Reversion Timer, Park Monitoring Periodic Reversion Timer, and Park Monitoring Forward No Retrieve Timer. Each service parameter includes a default and requires no special configuration. These timer parameters are for park monitoring only; the Call Park Display Timer and Call Park Reversion Timer are not used for park monitoring. See the following table for descriptions of these parameters.

Configure the timers in the Cisco Unified Communications Manager Service Parameters page.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **System** > **Service Parameters**.

Step 2 Update the Park Monitoring Reversion Timer, Park Monitoring Periodic Reversion Timer, and Park Monitoring Forward No Retrieve Timer fields in the Clusterwide Parameters (Feature-General) pane.

Table 35: Service Parameters for Park Monitoring

Field	Description
Park Monitoring Reversion Timer	Default is 60 seconds. This parameter determines the number of seconds that Cisco Unified Co Manager waits before prompting the user to retrieve a call that the user parked. This timer st user presses Park on the phone, and a reminder is issued when the timer expires. You can override the value that this service parameter specifies on a per-line basis in the Par section of the Directory Number Configuration window (in Cisco Unified Communications Administration, choose **Call Routing** > **Directory Number**). Specify a value of 0 to immed the periodic reversion interval that the Park Monitoring Periodic Reversion Timer service pa specifies. (See the description that follows.) For example, if this parameter is set to zero and Monitoring Periodic Reversion Timer is set to 15, the user is immediately prompted about th and every 15 seconds thereafter until the Park Monitoring Forward No Retrieve Timer (see th that follows) expires.
Park Monitoring Periodic Reversion Timer	Default is 30 seconds. This parameter determines the interval (in seconds) that Cisco Unifie Communications Manager waits before prompting the user again that a call is parked. To co parked call, the user can simply go off-hook during one of these prompts. Cisco Unified Co Manager continues to prompt the user about the parked call as long as the call remains park the time that the Park Monitoring Forward No Retrieve Timer (see the description that follo expires. Specify a value of 0 to disable periodic prompts about the parked call.

Field	Description
Park Monitoring Forward No Retrieve Timer	Default is 300 seconds. This parameter determines the number of seconds that park reminder no occur before the parked call forwards to the Park Monitoring Forward No Retrieve destination specified in the parker Directory Number Configuration window. (If no forward destination is p Cisco Unified Communications Manager Administration, the call returns to the line that parked This parameter starts when the time that the Park Monitoring Reversion Timer service parameter expires. When the Park Monitoring Forward No Retrieve Timer expires, the call is removed from forwards to the specified destination or returns to the parker line.

Set Park Monitoring Parameters for Directory Numbers

The Directory Number Configuration window contains a Park Monitoring area where you can configure the three parameters.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **Call Routing** > **Directory Number**.

Step 2 Set the park monitoring fields as described in the following table.

Table 36: Park Monitoring Parameters

Field	Description
Park Monitoring Forward No Retrieve Destination External	When the parkee is an external party, the call forwards to the specified destination in the parker Park Monitoring Forward No Retrieve Destination External parameter. If the Forward No Retrieve Destination External field value is empty, the parkee is redirected to the parker line.
Park Monitoring Forward No Retrieve Destination Internal	When the parkee is an internal party, the call forwards to the specified destination in the parker's Park Monitoring Forward No Retrieve Destination Internal parameter. If the Forward No Retrieve Destination Internal is empty, the parkee is redirected to the parker line.
Park Monitoring Reversion Timer	This parameter determines the number of seconds that Cisco Unified Communications Manager waits before prompting the user to retrieve a call that the user parked. This timer starts when the user presses Park on the phone, and a reminder is issued when the timer expires. Default: 60 seconds If you configure a nonzero value, this value overrides the value of this parameter set in the Service Parameters window. However, if you configure a value of 0 here, then the value in the Service Parameters window is used.

Set Up Park Monitoring for Hunt Lists

When a call that was routed via the hunt list is parked, the Hunt Pilot Park Monitoring Forward No Retrieve Destination parameter value is used (unless it is blank) when the Park Monitoring Forward No Retrieve Timer expires.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **Call Routing** > **Route/Hunt** > **Hunt Pilot**.

Step 2 Set the Hunt Pilot Park Monitoring Forward No Retrieve Destination parameter.

If the Hunt Pilot Park Monitoring Forward No Retrieve Destination parameter value is blank, the call forwards to the destination that is configured in the Directory Number Configuration window when the Park Monitoring Forward No Retrieve Timer expires.

Set Up the Audio and Video Port Range

Audio and video traffic can be sent to different RTP port ranges in order to improve Quality of Service (QoS).

The following fields control the port ranges in the Cisco Unified Communications Manager Administration:

- Audio ports
  - Start Media Port (default: 16384)
  - Stop Media Port (default: 32766)
- Video ports
  - Start Video (This is to set the video start port).
    - Minimum: 2048
    - Maximum: 65535
  - Stop Video (This is to set the video stop port)
    - Minimum: 2048
    - Maximum: 65535

The following rules apply when configuring the video port fields:

After the Start Video RTP Port and Stop Video RTP Port are configured, the phone uses ports within the video port range for video traffic. The audio traffic uses the media ports.

If the audio and video port ranges overlap, the overlapped ports carry both audio and video traffic. If the video port range is not configured correctly, the phone uses the configured audio ports for both audio and video traffic.

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

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select **Device** > **Device Settings** > **SIP Profile**

Step 2 Set the Start Media Port and Stop Media Port fields for the audio port range.

Step 3 Select **Save**.

Step 4 Select one of the following windows:

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

Step 5 Set the Start Video RTP Port and Stop Video RTP Port fields for the range of ports required.

The following rules apply when configuring the video port fields:

- The value in the Stop Video RTP Port field must be larger than the value in the Start Video RTP Port field.
- The difference between the Start Video RTP Port field and the Stop Video RTP Port field must be at least 16.

Step 6 Select **Save**.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Set up Cisco IP Manager Assistant

Cisco IP Manager Assistant (IPMA) provides call routing and other call management features to help managers and assistants handle phone calls more effectively.

IPMA services must be configured in Cisco Unified Communications Manager before you can access them. For detailed information on configuring IPMA, see the *Feature Configuration Guide for Cisco Unified Communications Manager*.

IPMA has three key components:

Manager

A manager has calls intercepted by the call routing service.

Assistant

An assistant handles calls on behalf of a manager.

Assistant Console

The assistant console is a desktop application assistants can use to perform tasks and manage most features.

IPMA supports two modes of operation: proxy line support and shared line support. Both modes support multiple calls per line for the manager. The IPMA service supports both proxy line and shared line support in a cluster.

In shared-line mode, the manager and assistant share a directory number and calls are handled on the shared line. Both the manager phone and the assistant phone ring when a call is received on the shared line. Shared-line mode does not support default assistant selection, assistant watch, call filtering or divert all calls.

If you configure Cisco IPMA in shared-line mode, the manager and assistant share a directory number; for example, 1701. The assistant handles calls for a manager on the shared directory number. When a manager receives a call on directory number 1701, both the manager phone and the assistant phone rings.

Not all IPMA features are available in shared-line mode including default assistant selection, assistant watch, call filtering, and divert all calls. An assistant cannot see or access these features on the Assistant Console application. The assistant phone does not have the softkey for the Divert All feature. The manager phone does not have the softkeys for Assistant Watch, Call Intercept, or Divert All feature.

In order to access shared-line support on user devices, you must first use Cisco Unified Communications Manager Administration to configure and start the Cisco IP Manager Assistant service.

In proxy-line mode, the assistant handles calls on behalf of a manager using a proxy number. Proxy-line mode supports all IPMA features.

When you configure Cisco IPMA in proxy-line mode, the manager and assistant do not share a directory number. The assistant handles calls for a manager using a proxy number. The proxy number is not the directory number for the manager. It is an alternate number chosen by the system and is used by an assistant to handle manager calls. In proxy-line mode, a manager and an assistant have access to all features that are available in IPMA, which include default assistant selection, assistant watch, call filtering, and divert all.

In order to access proxy-line support on user devices, you must first use Cisco Unified Communications Manager Administration to configure and start the Cisco IP Manager Assistant service.

You access IPMA features using softkeys and through Phone Services. The softkey template is configured in Cisco Unified Communications Manager. IPMA supports the following standard softkey templates:

Standard Manager

Supports manager for proxy mode.

Standard Shared Mode Manager

Supports manager for shared mode.

Standard Assistant

Supports assistant in proxy or in shared mode.

The following table describes the softkeys available in the softkey templates.

Table 37: IPMA Softkeys

Softkey	Call State	Description
Redirect	Ringing, Connected, On Hold	Divert the selected call to a pre-configured target.
Intercept	All states	Divert a call from the assistant's phone to the manager's phone and auto answer it.

Softkey	Call State	Description
Set Watch	All states	View the status of call handled by an assistant.
TransVM	Ringing, Connected, On Hold	Redirect the selected call to the manager's voice mail.
Divert All	All states	Divert all calls that are routed to the manager to a preconfigured target.

Note Intercept, Set Watch, and Divert All should only be configured for a manager phone in proxy line mode.

The following procedure is an overview of the steps required.

Procedure

Step 1 Configure the phones and users.
Step 2 Associate the phones to the users.
Step 3 Activate the Cisco IP Manager Assistant service in the Service Activation window.
Step 4 Configure system administration parameters.
Step 5 If required, configure IPMA clusterwide services parameters.
Step 6 (Optional) Configure the user CAPF profile
Step 7 (Optional) Configure the IPMA service parameters for security
Step 8 Stop and restart the IPMA service.
Step 9 Configure phone parameter, manager, and assistant settings, including the softkey templates.
Step 10 Configure Cisco Unified Communications Manager Assistant application.
Step 11 Configure dial rules.
Step 12 Install the Assistant Console application.
Step 13 Configure the manager and assistant console applications.

Set up Visual Voicemail

Visual Voicemail is configured for all Cisco IP Phones or to an individual user or group of users, from the Cisco Unified Communications Manager Administration.

Note For configuration information, see the Cisco Visual Voicemail documentation at http://www.cisco.com/c/en/us/support/unified-communications/visual-voicemail/model.html.

The Visual Voicemail client is not supported as a midlet on any of the Cisco IP Phone 8800 phones.

Procedure

Step 1 In Cisco Unified Communications Manager Administration, choose **Device** > **Device Settings** > **Phone Services**.

Step 2 Select **Add New** to create a new service for Visual Voicemail.

Step 3 In the IP Phone Service Configuration window, enter the following information in their respective fields:

- Service Name—Enter **`VisualVoiceMail`**.
- ASCII Service Name—Enter **`VisualVoiceMail`**.
- Service URL—Enter as **`Application:Cisco/VisualVoiceMail`**.
- Service Category—Select **XML Service** from the pulldown menu.
- Service Type—Select **Messages** from the pulldown menu.

Step 4 Check **Enable** and click **Save**.

Note Ensure that you do not check **Enterprise Subscription**.

Step 5 In the Service Parameter Information window, click **New Parameter** and enter the following information in their respective fields:

- Parameter Name. Enter voicemail_server.
- Parameter Display Name. Enter voicemail_server.
- Default Value. Enter the hostname of the primary Unity Server.
- Parameter Description

Step 6 Check **Parameter is Required** and click **Save**.

Note Ensure that you do not check **Parameter is a Password (mask contents)**.

Step 7 Close the window and select **Save** again in the Phone Service Configuration window.

Set Up Visual Voicemail for a Specific User

Use the following procedure to configure Visual Voicemail for a specific user.



Note For configuration information, see the Cisco Visual Voice Mail documentation at http://www.cisco.com/c/en/us/support/unified-communications/visual-voicemail/model.html.

Procedure

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

Step 2 Select the device associated to the user you are searching for.

Step 3 In the Related Links drop-down, choose **Subscribe/Unsubscribe Services** and click **Go**.

Step 4 Select the VisualVoiceMail Service you created, then choose **Next** > **Subscribe**.

Visual Voicemail Setup for a User Group

To add a batch of Cisco IP phones to Cisco Unified Communications Manager with Visual Voicemail subscribed, create a phone template in BAT tool for each phone type and in each phone template. You can then subscribe to the Visual Voicemail service, and use the template to insert the phones.

If you already have your Cisco IP phones registered and want to get phones subscribed to the Visual Voicemail service, create a phone template in BAT, subscribe to the Visual Voicemail service in the template, and then use BAT tool to update phones.

For more information, see http://www.cisco.com/c/en/us/support/unified-communications/visual-voicemail/model.html.

Assured Services SIP

Assured Services SIP(AS-SIP) is a collection of features and protocols that offer a highly secure call flow for Cisco IP Phones and third-party phones. The following features are collectively known as AS-SIP:

- Multilevel Precedence and Preemption (MLPP)
- Differentiated Services Code Point (DSCP)
- Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP)
- Internet Protocol version 6 (IPv6)

AS-SIP is often used with Multilevel Precedence and Preemption (MLPP) to prioritize calls during an emergency. With MLPP, you assign a priority level to your outgoing calls, from level 1 (low) to level 5 (high). When you receive a call, a precedence level icon displays on the phone that shows the call priority.

To configure AS-SIP, complete the following tasks on Cisco Unified Communications Manager:

- Configure a Digest User—Configure the end user to use digest authentication for SIP requests.
- Configure SIP Phone Secure Port—Cisco Unified Communications Manager uses this port to listen to SIP phones for SIP line registrations over TLS.
- Restart Services—After configuring the secure port, restart the Cisco Unified Communications Manager and Cisco CTL Provider services. Configure SIP Profile for AS-SIP-Configure a SIP profile with SIP settings for your AS-SIP endpoints and for your SIP trunks. The phone-specific parameters are not downloaded to a third-party AS-SIP phone. They are used only by Cisco Unified Manager. Third-party phones must locally configure the same settings.
- Configure Phone Security Profile for AS-SIP—You can use the phone security profile to assign security settings such as TLS, SRTP, and digest authentication.
- Configure AS-SIP Endpoint—Configure a Cisco IP Phone or a third-party endpoint with AS-SIP support.
- Associate Device with End Use—Associate the endpoint with a user.
- Configure SIP Trunk Security Profile for AS-SIP—You can use the sip trunk security profile to assign security features such as TLS or digest authentication to a SIP trunk.
- Configure SIP Trunk for AS-SIP—Configure a SIP trunk with AS-SIP support.
- Configure AS-SIP Features—Configure additional AS-SIP features such as MLPP, TLS, V.150, and IPv6.

For detailed information about configuring AS-SIP, see the "Configure AS-SIP Endpoints" chapter, *System Configuration Guide for Cisco Unified Communications Manager*.

Migration of your Phone to a Multiplatform Phone Directly

You can migrate your enterprise phone to a multiplatform phone easily in one step without using transition firmware load. All you need is to obtain and authorize the migration license from the server.

For more information, see https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cuipph/MPP/MPP-conversion/enterprise-to-mpp/cuip_b_conversion-guide-ipphone.html

Multilevel Precedence and Preemption

Multilevel Precedence and Preemption (MLPP) allows you to prioritize calls during emergencies or other crisis situations. You assign a priority to your outgoing calls that range from 1 to 5. Incoming calls display an icon that shows the call priority. Authenticated users can preempt calls either to targeted stations or through fully subscribed TDM trunks.

This capability assures high-ranking personnel of communication to critical organizations and personnel.

MLPP is often used with Assured Services SIP(AS-SIP). For detailed information about configuring MLPP, see the "Configure Multilevel Precedence and Preemption" chapter, *System Configuration Guide for Cisco Unified Communications Manager*.

Set Up Softkey Template

Using Cisco Unified Communications Manager Administration, you can associate a maximum of 18 softkeys with applications that are supported by the phone. Cisco Unified Communications Manager supports the Standard User and Standard Feature softkey template.

An application that supports softkeys has one or more standard softkey templates associated with it. You modify a standard softkey template by copying it, renaming it, and then updating the new template. You can also modify a nonstandard softkey template.

The Softkey Control parameter shows if softkeys of a phone are controlled by the Softkey Template feature. The Softkey Control parameter is a required field.

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

The Cisco IP Phones do not support all the softkeys that are configurable in Softkey Template Configuration on Cisco Unified Communications Manager Administration. Cisco Unified Communications Manager allows you to enable or disable some softkeys in the control policy configuration settings. The following table lists the features and the softkeys that can be configured on a softkey template, and identifies whether it is supported on the Cisco IP Phones.



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

Table 38: Configurable Softkeys

Feature	Configurable Softkeys in the Softkey Template configuration	Supported as a Softkey
Answer	Answer (Answer)	Supported
Call Back	Call Back (CallBack)	Supported
Call Forward All	Forward All (cfwdAll)	Supported
Call Park	Call Park (Park)	Supported
Call Pickup	Pick Up (Pickup)	Supported
Barge	Barge	Supported
cBarge	Conference Barge	Supported
Conference	Conference (Confrn)	Supported
Conference List	Conference List (ConfList)	Supported
Divert	Immediate Divert (iDivert)	Supported
Do Not Disturb	Toggle Do Not Disturb (DND)	Supported
End Call	End Call (EndCall)	Supported
Group Pickup	Group Pick UP (GPickUp)	Supported
Hold	Hold (Hold)	Supported
Hunt Group	HLog (HLog)	Supported
Join	Join (Join)	Not supported
Malicious Call Identification	Toggle Malicious Call Identification (MCID)	Supported
Meet Me	Meet Me (MeetMe)	Supported
Mobile Connect	Mobility (Mobility)	Supported
New Call	New Call (NewCall)	Supported
Other Pickup	Other Pickup (oPickup)	Supported
PLK Support for Queue Statistics	Queue Status	Not supported
Quality Reporting Tool	Quality Reporting Tool (QRT)	Supported
Redial	Redial (Redial)	Supported

Feature	Configurable Softkeys in the Softkey Template configuration	Supported as a Softkey
Remove Last Conference Participant	Remove Last Conference Participant (Remove)	Not supported
Resume	Resume (Resume)	Supported
Select	Select (Select)	Not supported
Speed Dial	Abbreviated Dial (AbbrDial)	Supported
Transfer	Transfer (Trfr)	Supported
Video Mode Command	Video Mode Command (VidMode)	Not supported

Procedure

Step 1 In Cisco Unified Communications Manager Administration, select one of the following windows:

- To configure softkey templates, select **Device** > **Device Settings** > **SoftkeyTemplate**.
- To assign a softkey template to a phone, select **Device** > **Phone** and configure the Softkey Template field.

Step 2 Save the changes.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Phone Button Templates

Phone button templates let you assign speed dials and call-handling features to programmable buttons. Call-handling features that can be assigned to buttons include Answer, Mobility, and All Calls.

Ideally, you modify templates before you register phones on the network. In this way, you can access customized phone button template options from Cisco Unified Communications Manager during registration.

Modify Phone Button Template

For more information about IP Phone services and configuring line buttons, see the documentation for your particular Cisco Unified Communications Manager release.

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **Device** > **Device Settings** > **Phone Button Template**.

Step 2 Click **Find**.

Step 3 Select the phone model.

Step 4 Select **Copy**, enter a name for the new template, and then select **Save**.

The Phone Button Template Configuration window opens.

Step 5 Identify the button that you would like to assign, and select **Service URL** from the Features drop-down list that associates with the line.

Step 6 Select **Save** to create a new phone button template that uses the service URL.

Step 7 Choose **Device** > **Phone** and open the Phone Configuration window for the phone.

Step 8 Select the new phone button template from the Phone Button Template drop-down list.

Step 9 Select **Save** to store the change and then select **Apply Config** to implement the change.

The phone user can now access the Self Care Portal and associate the service with a button on the phone.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Assign Phone Button Template for All Calls

Assign an All Calls button in the phone template for users with multiple shared lines.

When you configure an All Calls button on the phone, users use the All Calls button to:

- See a consolidated list of current calls from all lines on the phone.
- See (under Call History) a list of all missed calls from all lines on the phone.
- Place a call on the user's primary line when the user goes off-hook. All Calls automatically defaults to the user primary line for any outgoing call.

Procedure

Step 1 Modify the phone button template to include the All Calls button.

Step 2 Assign the template to the phone.

Set Up PAB or Speed Dial as IP Phone Service

You can modify a phone button template to associate a service URL with a programmable button. Doing so provides users with single-button access to the PAB and Speed Dials. Before you modify the phone button template, you must configure PAB or Speed Dials as an IP Phone service. For more information, see the documentation for your particular Cisco Unified Communications Manager release.

To configure PAB or Speed Dial as an IP Phone service (if it is not already a service), follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **Device** > **Device Settings** > **Phone Services**.

The Find and List IP Phone Services window displays.

Step 2 Click **Add New**.

The IP Phone Services Configuration window displays.

Step 3 Enter the following settings:

- Service Name: Enter **Personal Address Book**.
- Service Description: Enter an optional description of the service.
- Service URL

  For PAB, enter the following URL:

  http://<Unified CM-server-name>**:8080/ccmpd/login.do?name=#DEVICENAME#&service=pab**

  For Fast Dial, enter the following URL:

  http://<Unified-CM-server-name>**:8080/ccmpd/login.do?name=#DEVICENAME#&service=fd**
- Secure Service URL

  For PAB, enter the following URL:

  https://<Unified CM-server-name>**:8443/ccmpd/login.do?name=#DEVICENAME#&service=pab**

  For Fast Dial, enter the following URL:

  https://<Unified-CM-server-name>**:8443/ccmpd/login.do?name=#DEVICENAME#&service=fd**
- Service Category: Select **XML Service**.
- Service Type: Select **Directories**.
- Enable: Select the check box.

  http://<IP_address> or *https://<IP_address>* (Depends on the protocol that the Cisco IP Phone supports.)

Step 4 Select **Save**.

Note If you change the service URL, remove an IP Phone service parameter, or change the name of a phone service parameter for a phone service to which users are subscribed, you must click **Update Subscriptions** to update all currently subscribed users with the changes; otherwise, users must resubscribe to the service to rebuild the correct URL.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Modify Phone Button Template for PAB or Fast Dial

You can modify a phone button template to associate a service URL with a programmable button. Doing so provides users with single-button access to the PAB and Speed Dials. Before you modify the phone button template, you must configure PAB or Speed Dials as an IP Phone service.

For more information about IP Phone services and configuring line buttons, see the documentation for your particular Cisco Unified Communications Manager release.

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **Device** > **Device Settings** > **Phone Button Template**.

Step 2 Click **Find**.

Step 3 Select the phone model.

Step 4 Select **Copy**, enter a name for the new template, and then select **Save**.

The Phone Button Template Configuration window opens.

Step 5 Identify the button that you would like to assign, and select **Service URL** from the Features drop-down list that associates with the line.

Step 6 Select **Save** to create a new phone button template that uses the service URL.

Step 7 Choose **Device** > **Phone** and open the Phone Configuration window for the phone.

Step 8 Select the new phone button template from the Phone Button Template drop-down list.

Step 9 Select **Save** to store the change and then select **Apply Config** to implement the change.

The phone user can now access the Self Care Portal and associate the service with a button on the phone.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

VPN Configuration

The Cisco VPN feature helps you to preserve network security while giving users a safe, reliable method to connect to your corporate network. Use this feature when:

- A phone is located outside a trusted network
- Network traffic between the phone and Cisco Unified Communications Manager crosses an untrusted network

With a VPN, there are three common approaches to client authentication:

- Digital certificates
- Passwords
- Username and password

Each method has its advantages. But if your corporate security policy permits it, we recommend a certificate-based approach because certificates allow for a seamless sign-in without any user intervention. Both LSC and MIC certificates are supported.

To configure any of the VPN features, provision the device on-premises first and then you can deploy the device off-premise.

For more information about certification authentication and working with VPN network, see the Technical Note *AnyConnect VPN Phone with Certificate Authentication on an ASA Configuration Example*. The URL for this document is http://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/115785-anyconnect-vpn-00.html.

With a password, or username and password approach a user is prompted for sign-in credentials. Set the user sign-in credentials in accordance with your company security policy. You can also configure the Enable Password Persistence setting so that the user password is saved on the phone. The user password is saved until either a failed log-in attempt occurs, a user manually clears the password, or the phone resets or loses power.

Another useful tool is the Enable Auto Network Detection setting. When you enable this check box, the VPN client can only run when it detects that it is outside the corporate network. This setting is disabled by default.

Your Cisco phone supports Cisco SVC IPPhone Client v1.0 as the client type.

For additional information about maintaining, configuring, and operating a virtual private network with a VPN, see *Security Guide for Cisco Unified Communications Manager*, "Virtual Private Network Setup" chapter. The URL for this document is http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

The Cisco VPN feature uses Secure Sockets Layer (SSL) to preserve network security.

Note Enter the Alternate TFTP server setting when you're configuring an off-premises phone for SSL VPN to ASA using a built-in client.

Set Up Additional Line Keys

Enable Enhanced line mode to use the buttons on both sides of the phone screen as line keys. Predictive dialing and Actionable incoming call alerts are enabled by default in Enhanced line mode.

Before you begin

You must create a new, customized phone button template.

Procedure

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

Step 2 Locate the phone that you need to set up.

Step 3 Navigate to the Product Specific Configuration area and set the **Line Mode** field to **Enhanced Line Mode**.

Step 4 Navigate to the Device Information area and set the **Phone Button Template** field to a customized template.

Step 5 Select **Apply Config**.

Step 6 Select **Save**.

Step 7 Restart the phone.

Related Topics

Session Line Mode Environment, on page 155

Features Available in Enhanced Line Mode

Enhanced line mode (ELM) can be used with Mobile and Remote Access Through Expressway.

ELM can also be used with a rollover line, a call routing configuration in which calls are forwarded to another shared line if the initial shared line is busy. When ELM is used with a rollover line, recent calls to shared lines are consolidated under a single directory number. For more information about rollover lines, see *Feature Configuration Guide for Cisco Unified Communications Manager* for Cisco Unified Communications Manager 12.0(1) or later.

ELM supports most but not all features. Enabling a feature does not imply support. Read the following table to confirm that a feature is supported.

Table 39: Feature Support and Enhanced Line Mode

Feature	Supported	Firmware Release
Answer	Yes	11.5(1) and later
Automatically Answer Calls	Yes	11.5(1) and later
Barge/cBarge	Yes	11.5(1) and later
Directed Call Park with BLF	Yes	12.0(1) and later
Bluetooth Smartphone integration	No	-
Bluetooth USB Headsets	Yes	11.5(1) and later
Call Back	Yes	11.5(1) and later
Call Chaperone	No	-
Call Forward All	Yes	11.5(1) and later
Call Park	Yes	12.0(1) and later
Call Park Line Status	Yes	12.0(1) and later
Call Pickup	Yes	11.5(1) and later
Call Pickup Line Status	Yes	11.5(1) and later
Call Forward All on Multiple Lines	Yes	11.5(1) and later
Cisco Extension Mobility Cross Cluster	Yes	12.0(1) and later supports this feature.

Feature	Supported	Firmware Release
Cisco IP Manager Assistant (IPMA)	No	-
Cisco Unified Communications Manager Express	No	-
Conference	Yes	11.5(1) and later
Computer Telephony Integration (CTI) applications	Yes	11.5(1) and later
Decline	Yes	11.5(1) and later
Device Invoked Recording	Yes	11.5(1)SR1 and later
Do Not Disturb	Yes	11.5(1) and later
Enhanced SRST	No	-
Extension Mobility	Yes	11.5(1) and later
Group Pickup	Yes	12.0(1) and later supports this feature.
Hold	Yes	11.5(1) and later
Hunt Groups	Yes.	12.0(1) and later
Incoming Call Alert with configurable timer	No	-
Intercom	Yes	11.5(1) and later
Key Expansion Module	Cisco IP Phone 8851/8861 Key Expansion Module and Cisco IP Phone 8865 Key Expansion Module support Enhanced Line Mode	12.0(1) and later
Malicious Call Identification (MCID)	Yes	11.5(1) and later
Meet Me	Yes	11.5(1) and later
Mobile Connect	Yes	11.5(1) and later
Multilevel Precedence and Preemption	No	-
Mute	Yes	11.5(1) and later
Other PickUp	Yes	12.0(1) and later

Feature	Supported	Firmware Release
Programmable Line Key (PLK) Support for Queue Status	Yes	11.5(1) and later
Privacy	Yes	11.5(1) and later
Queue Status	Yes	11.5(1) and later
Quality Reporting Tool (QRT)	Yes	11.5(1) and later
Right to Left locale support	No	-
Redial	Yes	11.5(1) and later
Silent Monitoring and Recording	Yes	11.5(1)SR1 and later
Speed Dial	Yes	11.5(1) and later
Survivable Remote Site Telephony (SRST)	Yes	11.5(1) and later
Transfer	Yes	11.5(1) and later
Uniform Resource Identifier (URI) Dialing	Yes	11.5(1) and later
Video Calls	Yes	11.5(1) and later
Visual Voicemail	Yes	11.5(1) and later
Voicemail	Yes	11.5(1) and later

Related Topics

Session Line Mode Environment, on page 155

Set Up TLS Resumption Timer

TLS Session resumption enables a TLS session to resume without repeating the entire TLS authentication process. It can significantly reduce the time taken for TLS connection to exchange data.

Although the phones support TLS sessions, all TLS sessions do not support TLS resumption. The following list describes the different sessions and TLS resumption support:

- TLS session for SIP signaling: supports resumption
- HTTPs client: supports resumption
- CAPF: supports resumption
- TVS: supports resumption
- EAP-TLS: does not support resumption
- EAP-FAST: does not support resumption

- VPN client: does not support resumption

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

Procedure

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

Step 2 Set the TLS Resumption Timer parameter.

The range for the timer is 0 to 3600 sec. The default value is 3600. If the field is set to 0, then TLS session resumption is disabled.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Enable Intelligent Proximity

Note This procedure only applies to Bluetooth enabled phones. Cisco IP Phone 8811, 8841, 8851NR, and 8865NR do not support Bluetooth.

Intelligent Proximity enables users to take advantage of the acoustic properties of the phone with their mobile device or tablet. The user pairs the mobile device or tablet to the phone using Bluetooth.

With a mobile device paired, the user can place and receive mobile calls on the phone. With a tablet, the user can route the audio from the tablet to the phone.

Users can pair multiple mobile devices, tablets, and a Bluetooth headset to the phone. However, only one device and one headset can be connected at the same time.

Procedure

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

Step 2 Locate the phone you want to modify.

Step 3 Locate the Bluetooth field and set the field to **Enabled**.

Step 4 Locate the Allow Bluetooth Mobile Handsfree Mode field, and set the field to **Enabled**.

Step 5 Save the changes and apply them to the phone.

Video Transmit Resolution Setup

Cisco IP Phone 8845, 8865, and 8865NR supports the following video formats:

- 720p (1280x720)
- WVGA(800x480)
- 360p (640x360)
- 240p (432x240)
- VGA (640x480)
- CIF (352x288)
- SIF (352x240)
- QCIF (176x144)

Cisco IP Phones with video capacity negotiate best resolution for bandwidth based on phone configures or resolution limitations. Example: On a direct 88x5 to 88x5 call, the phones do not send true 720p, they send 800x480. This limitation is purely due to the 5-inch WVGA screen resolution on the 88x5 being 800 x 480.

Video type	Video resolution	Frames per second (fps)	Video bit rate range
720p	1280 x 720	30	1360–2500 kbps
720p	1280 x 720	15	790–1359 kbps
WVGA	800 x 480	30	660–789 kbps
WVGA	800 x 480	15	350–399 kbps
360p	640 x 360	30	400–659 kbps
360p	640 x 360	15	210–349kbps
240p	432 x 240	30	180–209kbps
240p	432 x 240	15	64–179kbps
VGA	640 x 480	30	520–1500kbps
VGA	640 x 480	15	280–519kbps
CIF	352 x 288	30	200–279 kbps
CIF	352 x 288	15	120–199 kbps
SIF	352 x 240	30	200–279 kbps
SIF	352 x 240	15	120–199 kbps
QCIF	176 x 144	30	94–119 kbps
QCIF	176 x 144	15	64–93 kbps

Headset Management on Older Versions of Cisco Unified Communications Manager

If you have a version of Cisco Unified Communications Manager older than 12.5(1)SU1, you can remotely configure your Cisco headset settings for use with on-premises phones.

Remote headset configuration on Cisco Unified Communication Manager version 10.5(2), 11.0(1), 11.5(1), 12.0(1), and 12.5(1) requires you to download a file from the Cisco Software Download website, edit the file, and then upload the file on the Cisco Unified Communications Manager TFTP server. The file is a JavaScript Object Notification (JSON) file. The updated headset configuration is applied to the enterprise headsets over a 10 to 30-minute time frame to prevent a traffic backlog on the TFTP server.

Note You can manage and configure headsets through Cisco Unified Communications Manager Administration version 11.5(1)SU7.

Note the following as you work with the JSON file:

- The settings aren't applied if you are missing a bracket or brackets in the code. Use an online tool such as JSON Formatter and check the format.
- Set the **updatedTime** setting to the current epoch time or the configuration is not applied. Alternatively, you can increase the **updatedTime** value by +1 to make it larger than the previous version.
- Do not change the parameter name or the setting will not be applied.

For more information on the TFTP service, see the "Manage Device Firmware" chapter of the *Administration Guide for Cisco Unified Communications Manager and IM and Presence Service*.

Upgrade your phones to the latest firmware release before you apply the `defaultheadsetconfig.json` file. The following table describes the default settings you can adjust with the JSON file.

Download the Default Headset Configuration File

Before configuring the headset parameters remotely, you must download the latest JavaScript Object Notation (JSON) sample file.

Procedure

Step 1 Go to the following URL: https://software.cisco.com/download/home/286320550.

Step 2 Choose **Headsets 500 Series**.

Step 3 Select your headset series.

Step 4 Choose a release folder and select the zip file.

Step 5 Click the **Download** or **Add to cart** button, and follow the prompts.

Step 6 Unzip the file to a directory on your PC.

What to do next

Modify the Default Headset Configuration File, on page 196

Modify the Default Headset Configuration File

Note the following as you work with the JavaScript Object Notation (JSON) file:

- The settings aren't applied if you are missing a bracket or brackets in the code. Use an online tool such as JSON Formatter and check the format.
- Set the "**updatedTime**" setting to the current epoch time or the configuration is not applied.
- Confirm that **firmwareName** is `LATEST` or the configurations will not be applied.
- Do not change a parameter name or the setting will not be applied.

Procedure

Step 1 Open the `defaultheadsetconfig.json` file with a text editor.

Step 2 Edit the **updatedTime** and the headset parameter values you wish to modify.

A sample script is shown below. This script is provided for reference only. Use it as a guide as you configure your headset parameters. Use the JSON file that was included with your firmware load.

```
{
  "headsetConfig": {
    "templateConfiguration": {
      "configTemplateVersion": "1",
      "updatedTime": 1537299896,
      "reportId": 3,
      "modelSpecificSettings": [
        {
          "modelSeries": "530",
          "models": [
            "520",
            "521",
            "522",
            "530",
            "531",
            "532"
          ],
          "modelFirmware": [
            {
              "firmwareName": "LATEST",
              "latest": true,
              "firmwareParams": [
                {
                  "name": "Speaker Volume",
                  "access": "Both",
                  "usageId": 32,
                  "value": 7
                },
                {
                  "name": "Microphone Gain",
                  "access": "Both",
                  "usageId": 33,
                  "value": 2
                },
```

```
        {
          "name": "Sidetone",
          "access": "Both",
          "usageId": 34,
          "value": 1
        },
        {
          "name": "Equalizer",
          "access": "Both",
          "usageId": 35,
          "value": 3
        }
      ]
    }
  ]
},
{
  "modelSeries": "560",
  "models": [
    "560",
    "561",
    "562"
  ],
  "modelFirmware": [
    {
      "firmwareName": "LATEST",
      "latest": true,
      "firmwareParams": [
        {
          "name": "Speaker Volume",
          "access": "Both",
          "usageId": 32,
          "value": 7
        },
        {
          "name": "Microphone Gain",
          "access": "Both",
          "usageId": 33,
          "value": 2
        },
        {
          "name": "Sidetone",
          "access": "Both",
          "usageId": 34,
          "value": 1
        },
        {
          "name": "Equalizer",
          "access": "Both",
          "usageId": 35,
          "value": 3
        },
        {
          "name": "Audio Bandwidth",
          "access": "Admin",
          "usageId": 36,
          "value": 0
        },
        {
          "name": "Bluetooth",
          "access": "Admin",
          "usageId": 39,
          "value": 0
        },
```

```
                {
                  "name": "DECT Radio Range",
                  "access": "Admin",
                  "usageId": 37,
                  "value": 0
                }
                {
                   "name": "Conference",
                  "access": "Admin",
                  "usageId": 41,
                  "value": 0
              ]
            }
          ]
        }
      ]
    }
  }
}
```

Step 3 Save the `defaultheadsetconfig.json`.

What to do next

Install the default configuration file.

Install the Default Configuration File on Cisco Unified Communications Manager

After you edit the `defaultheadsetconfig.json` file, install it on Cisco Unified Communications Manager using the TFTP File Management tool.

Procedure

Step 1 From Cisco Unified OS Administration, choose **Software Upgrades** > **TFTP File Management**.

Step 2 Select **Upload File**.

Step 3 Select **Choose File** and navigate to the `defaultheadsetconfig.json` file.

Step 4 Select **Upload File**.

Step 5 Click **Close**.

Restart the Cisco TFTP Server

After you upload the `defaultheadsetconfig.json` file to the TFTP directory, restart the Cisco TFTP server and reset the phones. After about 10–15 minutes, the download process begins and the new configurations are applied to the headsets. It takes an additional 10 to 30 minutes for the settings to be applied.

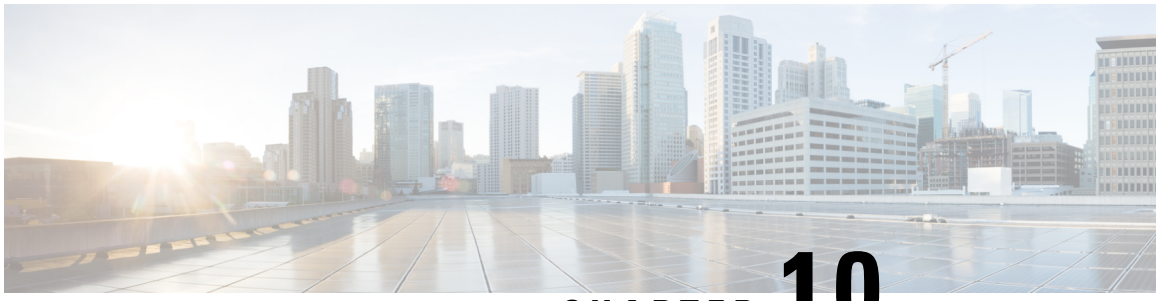
Procedure

Step 1 Log in to Cisco Unified Serviceability and choose **Tools** > **Control Center - Feature Services**.

Step 2 From the **Server** drop-down list box, choose the server on which the Cisco TFTP service is running.

Step 3 Click the radio button that corresponds to the **Cisco TFTP** service.

Step 4 Click **Restart**.



CHAPTER 10

Corporate and Personal Directory

- Corporate Directory Setup, on page 201
- Personal Directory Setup, on page 201
- User Personal Directory Entries Setup, on page 202

Corporate Directory Setup

The Corporate Directory allows a user to look up phone numbers for coworkers. To support this feature, you must configure corporate directories.

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

Cisco IP Phones use dynamic allocation for SecureApp on both client and servers. This ensures your phone can read certificates larger than 4KB, and reduces the frequency of `Host Not Found` error messages when a user accesses their directory.

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

After you complete the LDAP directory configuration, users can use the Corporate Directory service on their phone to look up users in the corporate directory.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Personal Directory Setup

The Personal Directory allows a user to store a set of personal numbers.

Personal Directory consists of the following features:

- Personal Address Book (PAB)
- Speed Dials
- Address Book Synchronization Tool (TABSynch)

Users can use these methods to access Personal Directory features:

- From a web browser—Users can access the PAB and Speed Dials features from the Cisco Unified Communications Self Care Portal.
- From the CiscoIP Phone—Choose **Contacts** to search the corporate directory or the user personal directory.
- From a Microsoft Windows application—Users can use the TABSynch tool to synchronize their PABs with Microsoft Windows Address Book (WAB). Customers who want to use the Microsoft Outlook Address Book (OAB) should begin by importing the data from the OAB into the WAB. TabSync can then be used to synchronize the WAB with Personal Directory. For instructions about TABSync, see Download Cisco IP Phone Address Book Synchronizer, on page 202 and Set Up Synchronizer, on page 203.

Cisco IP Phones use dynamic allocation for SecureApp on both client and servers. This ensures your phone can read certificates larger than 4KB, and reduces the frequency of `Host Not Found` error messages when a user accesses their directory.

To ensure that Cisco IP Phone Address Book Synchronizer users access only their end-user data, activate the Cisco UXL Web Service in Cisco Unified Serviceability.

To configure Personal Directory from a web browser, users must access their Self Care Portal. You must provide users with a URL and sign-in information.

User Personal Directory Entries Setup

Users can configure personal directory entries on the Cisco IP Phone. To configure a personal directory, users must have access to the following:

- Self Care Portal: Make sure that users know how to access their Self Care Portal. See Set Up User Access to the Self Care Portal, on page 77 for details.
- Cisco IP Phone Address Book Synchronizer: Make sure to provide users with the installer. See Download Cisco IP Phone Address Book Synchronizer, on page 202.

Note The Cisco IP Phone Address Book Synchronizer is only supported on unsupported versions of Windows (for example, Windows XP and earlier). The tool is not supported on newer versions of Windows. In future, it will be removed from the Cisco Unified Communications Manager plug-ins list.

Download Cisco IP Phone Address Book Synchronizer

To download a copy of the synchronizer to send to your users, follow these steps:

Procedure

Step 1 To obtain the installer, choose **Application** > **Plugins** from Cisco Unified Communications Manager Administration.

Step 2 Select **Download**, which is located next to the Cisco IP Phone Address Book Synchronizer plugin name.

Step 3 When the file download dialog box displays, select **Save**.

Step 4 Send the TabSyncInstall.exe file and the instructions in Cisco IP Phone Address Book Synchronizer Deployment, on page 203 to all users who require this application.

Cisco IP Phone Address Book Synchronizer Deployment

The Cisco IP Phone Address Book Synchronizer synchronizes data that is stored in your Microsoft Windows address book with the Cisco Unified Communications Manager directory and the Self Care Portal Personal Address Book.

Tip To successfully synchronize the Windows address book with the Personal Address Book, all Windows address book users should be entered in the Windows address book before you perform the following procedures.

Install Synchronizer

To install the Cisco IP Phone Address Book Synchronizer, follow these steps:

Procedure

Step 1 Get the Cisco IP Phone Address Book Synchronizer installer file from your system administrator.

Step 2 Double-click the TabSyncInstall.exe file that your administrator provided.

Step 3 Select **Run**.

Step 4 Select **Next**.

Step 5 Read the license agreement information, and select the **I Accept**. Select **Next**.

Step 6 Choose the directory in which you want to install the application and select **Next**.

Step 7 Select **Install**.

Step 8 Select **Finish**.

Step 9 To complete the process, follow the steps in Set Up Synchronizer, on page 203.

Set Up Synchronizer

To configure the Cisco IP Phone Address Book Synchronizer, perform these steps:

Procedure

Step 1 Open the Cisco IP Phone Address Book Synchronizer.

If you accepted the default installation directory, you can open the application by choosing **Start** > **All Programs** > **Cisco Systems** > **TabSync**.

Step 2 To configure user information, select **User**.

Step 3 Enter the Cisco IP Phone user name and password and select **OK**.

Step 4 To configure Cisco Unified Communications Manager server information, select **Server**.

Step 5 Enter the IP address or host name and the port number of the Cisco Unified Communications Manager server and select **OK**.

If you do not have this information, contact your system administrator.

Step 6 To start the directory synchronization process, select **Synchronize**.

The Synchronization Status window provides the status of the address book synchronization. If you chose the user intervention for duplicate entries rule and you have duplicate address book entries, the Duplicate Selection window displays.

Step 7 Choose the entry that you want to include in your Personal Address Book and select **OK**.

Step 8 When synchronization is complete, select **Exit** to close the Cisco Unified CallManager Address Book Synchronizer.

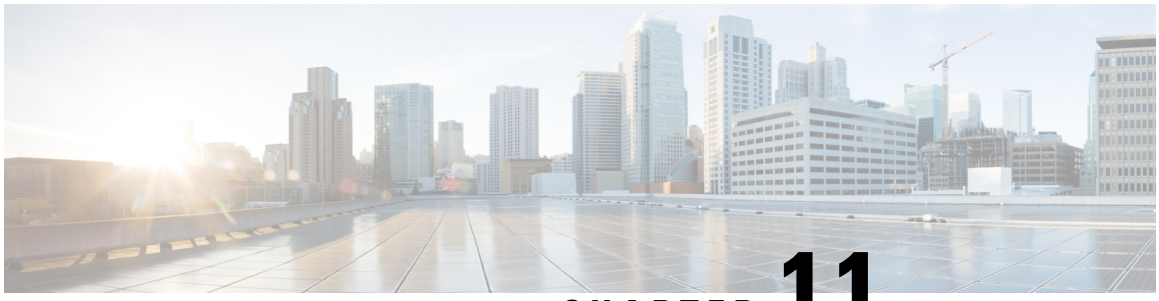
Step 9 To verify whether the synchronization worked, sign in to your Self Care Portal and choose **Personal Address Book**. The users from your Windows address book should be listed.



PART IV

Cisco IP Phone Troubleshooting

- Monitoring Phone Systems, on page 207
- Troubleshooting, on page 241
- Maintenance, on page 259
- International User Support, on page 265



CHAPTER 11

Monitoring Phone Systems

- Cisco IP Phone Status, on page 207
- Cisco IP Phone Web Page, on page 222
- Request Information from the Phone in XML, on page 237

Cisco IP Phone Status

This section describes how to view model information, status messages, and network statistics on the Cisco IP Phone 8800 series.

- Model Information: Displays hardware and software information about the phone.
- Status menu: Provides access to screens that display the status messages, network statistics, and statistics for the current call.

You can use the information that displays on these screens to monitor the operation of a phone and to assist with troubleshooting.

You can also obtain much of this information, and obtain other related information, remotely through the phone web page.

For more information about troubleshooting, see Troubleshooting, on page 241.

Display Phone Information Window

To display the Model Information screen, follow these steps.

Procedure

Step 1 Press **Applications** .

Step 2 Select **Phone Information**.

If the user is connected to a secure or authenticated server, a corresponding icon (lock or certificate) displays in the Phone Information Screen to the right of the server option. If the user is not connected to a secure or authenticated server, no icon appears.

Step 3 To exit the Model Information screen, press **Exit**.

Phone Information Fields

The following table describes the phone information settings.

Table 40: Phone Information Settings

Option	Description
Model number	Model number of the phone.
IPv4 Address	IP address of the phone.
Host name	Host name of the phone.
Active load	Version of firmware currently installed on the phone. The user can press **Details** for more information.
Inactive load	Inactive load appears only when a download is in progress. A download icon and a status of "Upgrade in Progress" or "Upgrade Failed" also display. If a user presses **Details** during an upgrade, the download filename and components are listed. A new firmware image can be set to download in advance of a maintenance window. Thus instead of waiting for all of the phones to download the firmware, the system switches more rapidly between resetting an existing load to Inactive status and installing the new load. When the download is complete, the icon changes to indicate the completed status; and a check mark displays for a successful download, or an "X" displays for a failed download. If possible, the rest of the loads continue to download.
Last upgrade	Date of the most recent firmware upgrade.
Active server	Domain name of the server to which the phone is registered.
Stand-by server	Domain name of the standby server.

Display Status Menu

The Status menu includes the following options, which provide information about the phone and phone operations:

- Status Messages: Displays the Status Messages screen, which shows a log of important system messages.
- Ethernet Statistic: Displays the Ethernet Statistics screen, which shows Ethernet traffic statistics.
- Wireless Statistics: Displays the Wireless Statistics screen, if applicable.
- Call Statistics: Displays counters and statistics for the current call.

- Current Access Point: Displays the Current Access Point screen, if applicable.

To display the Status menu, perform these steps:

Procedure

Step 1 To display the Status menu, press **Applications**.
Step 2 Select **Admin settings** > **Status**.
Step 3 To exit the Status menu, press **Exit**.

Display Status Messages Window

The Status Messages window displays the 30 most recent status messages that the phone has generated. You can access this screen at any time, even if the phone has not finished starting up.

Procedure

Step 1 Press **Applications**.
Step 2 Select **Admin settings** > **Status** > **Status messages**.
Step 3 To remove the current status messages, press **Clear**.
Step 4 To exit the Status Messages screen, press **Exit**.

Status Messages Fields

The following table describes the status messages that display on the Status Messages screen of the phone.

Table 41: Status Messages on the Cisco Unified IP Phone

Message	Description	Possible explanation and acti
CFG TFTP Size Error	The configuration file is too large for file system on the phone.	Power cycle the phone.
Checksum Error	Downloaded software file is corrupted.	Obtain a new copy of the phon the TFTPPath directory. You sl this directory when the TFTP s otherwise, the files may be cor
Could not acquire an IP address from DHCP	The phone has not previously obtained an IP address from a DHCP Server. This can occur when you perform an out of box or factory reset.	Confirm that the DHCP server address is available for the pho
CTL and ITL installed	The CTL and ITL files are installed on the phone.	None. This message is informa file nor the ITL file was install

Message	Description	Possible explanation and action
CTL Installed	A certificate trust list (CTL) file is installed in the phone.	None. This message is informatio was not installed previously.
CTL update failed	The phone could not update the certificate trust list (CTL) file.	Problem with the CTL file on the
DHCP timeout	DHCP server did not respond.	Network is busy: The errors should the network load reduces. No network connectivity between phone: Verify the network connect DHCP server is down: Check confi Errors persist: Consider assigning
DNS timeout	DNS server did not respond.	Network is busy: The errors should the network load reduces. No network connectivity between phone: Verify the network connect DNS server is down: Check configu
DNS unknown host	DNS could not resolve the name of the TFTP server or Cisco Unified Communications Manager.	Verify that the host names of the T Unified Communications Manage in DNS. Consider using IP addresses rathe
Duplicate IP	Another device is using the IP address that is assigned to the phone.	If the phone has a static IP address assigned a duplicate IP address. If you are using DHCP, check the configuration.
Erasing CTL and ITL files	Erasing CTL or ITL file.	None. This message is informatio
Error update locale	One or more localization files could not be found in the TFTPPath directory or were not valid. The locale was not changed.	From Cisco Unified Operating Syst that the following files are located the TFTP File Management: • Located in subdirectory with locale: • tones.xml • Located in subdirectory with s • glyphs.xml • dictionary.xml • kate.xml

Message	Description	Possible explanation and acti
File not found <Cfg File>	The name-based and default configuration file was not found on the TFTP Server.	The configuration file for a pho is added to the Cisco Unified C database. If the phone does not Communications Manager data generates a **CFG File Not** • Phone is not registered wi Communications Manage You must manually add th Communications Manage phones to autoregister. Se on page 68 for details. • If you are using DHCP, ve is pointing to the correct T • If you are using static IP ad of the TFTP server.
File Not Found <CTLFile.tlv>	This message displays on the phone when the Cisco Unified Communications Manager cluster is not in secure mode.	No impact; the phone can still Communications Manager.
IP address released	The phone is configured to release the IP address.	The phone remains idle until it reset the DHCP address.
ITL installed	The ITL file is installed in the phone.	None. This message is informat not installed previously.
Load rejected HC	The application that was downloaded is not compatible with the phone hardware.	Occurs if you attempted to inst this phone that did not support phone. Check the load ID that is assign Unified Communications Mana **Phone**). Reenter the load that c
No default router	DHCP or static configuration did not specify a default router.	If the phone has a static IP add router is configured. If you are using DHCP, the DH a default router. Check the DH
No DNS server IP	A name was specified but DHCP or static IP configuration did not specify a DNS server address.	If the phone has a static IP add server is configured. If you are using DHCP, the DH a DNS server. Check the DHC
No Trust List installed	The CTL file or the ITL file is not installed on the phone.	The trust list is not configured Communications Manager, whi by default.

Message	Description	Possible explanation and action
Phone failed to register. Cert key size is not FIPS compliant.	FIPS requires that the RSA server certificate is 2048 bits or greater.	Update the certificate.
Restart requested by Cisco Unified Communications Manager	The phone is restarting due to on a request from Cisco Unified Communications Manager.	Configuration changes were likely Cisco Unified Communications M pressed so that the changes take ef
TFTP access error	TFTP server is pointing to a directory that does not exist.	If you are using DHCP, verify that pointing to the correct TFTP serve If you are using static IP addresses TFTP server.
TFTP error	The phone does not recognize an error code that the TFTP server provided.	Contact Cisco TAC.
TFTP timeout	TFTP server did not respond.	Network is bus: The errors should r the network load reduces. No network connectivity between phone: Verify the network connect TFTP server is down: Check confi
Timed Out	Supplicant attempted 802.1X transaction but timed out to due the absence of an authenticator.	Authentication typically times out configured on the switch.
Trust List update failed	Update of the CTL and ITL files failed.	Phone has CTL and ITL files install the new CTL and ITL files. Possible reasons for failure: • Network failure occurred. • TFTP server was down. • The new security token that w and the TFTP certificate that are introduced, but are not ava and ITL files in the phone. • Internal phone failure occurre Possible solutions: • Check network connectivity. • Check whether the TFTP serv functioning normally. • If the Transactional Vsam Ser supported on Cisco Unified Co check whether the TVS server normally. • Verify whether the security to are valid. Manually delete the CTL and ITL solutions fail; reset the phone.

Message	Description	Possible explanation and acti
Trust List updated	The CTL file, the ITL file, or both files are updated.	None. This message is informa
Version error	The name of the phone load file is incorrect.	Make sure that the phone load
XmlDefault.cnf.xml, or .cnf.xml corresponding to the phone device name	Name of the configuration file.	None. This message indicates t file for the phone.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Display Network Information Screen

Use the information displayed on the Network Info screen to resolve connection issues on a phone.

A message is displayed on the phone if a user has trouble connecting to a phone network.

Procedure

Step 1 To display the Status menu, press **Applications** .

Step 2 Select **Admin settings** > **Status** > **Status messages**.

Step 3 Select **Network Info**.

Step 4 To exit Network Info, press **Exit**.

Display Network Statistics Screen

The Networks Statistics screen displays information about the phone and network performance.

To display the Network Statistics screen, follow these steps:

Procedure

Step 1 Press **Applications** .

Step 2 Select **Admin settings**>**Status**>**Network statistics**.

Step 3 To reset the Rx Frames, Tx Frames, and Rx Broadcasts statistics to 0, press **Clear**.

Step 4 To exit the Ethernet Statistics screen, press **Exit**.

Ethernet Statistics Information

The following tables describe the information in the Ethernet Statistics screen.

Table 42: Ethernet Statistics Information

Item	Description
Rx Frames	Number of packets that the phone received.
Tx Frames	Number of packets that the phone sent.
Rx Broadcasts	Number of broadcast packets that the phone received.
Restart Cause	Cause of the last reset of the phone. Specifies one of the following values: • Initialized • TCP-timeout • CM-closed-TCP • TCP-Bad-ACK • CM-reset-TCP • CM-aborted-TCP • CM-NAKed • KeepaliveTO • Failback • Phone-Keypad • Phone-Re-IP • Reset-Reset • Reset-Restart • Phone-Reg-Rej • Load Rejected HC • CM-ICMP-Unreach • Phone-Abort
Elapsed Time	Amount of time that has elapsed since the phone last rebooted.
Port 1	Link state and connection of the Network port. For example, **`Auto 100 Mb Full-Duplex`** means that the Network port is in a link-up state and has autonegotiated a full-duplex, 100-Mbps connection.
Port 2	Link state and connection of the PC port.
DHCP state (IPv4 / IPv6)	• In IPv4-only mode, displays only the DHCPv4 state, such as DHCP BOUND. • In IPv6-mode, displays only the DHCPv6 state, such as ROUTER ADVERTISE. • DHCPv6 state information is displayed.

The following tables describe the messages that appear for DHCPv4 and DHCPv6 states.

Table 43: DHCPv4 ethernet statistics messages

DHCPv4 state	Description
CDP INIT	CDP is not bound or WLAN is not in service
DHCP BOUND	DHCPv4 is BOUND
DHCP DISABLED	DHCPv4 is disabled
DHCP INIT	DHCPv4 is INIT
DHCP INVALID	DHCPv4 is INVALID; this is initial state
DHCP RENEWING	DHCPv4 is RENEWING
DHCP REBINDING	DHCPv4 is REBINDING
DHCP REBOOT	DHCPv4 is init-reboot
DHCP REQUESTING	DHCPv4 is requesting
DHCP RESYNC	DHCPv4 is RESYNCH
DHCP WAITING COLDBOOT TIMEOUT	DHCPv4 is booting
DHCP UNRECOGNIZED	Unrecognized DHCPv4 state
DISABLED DUPLICATE IP	Duplicated IPv4 Address
DHCP TIMEOUT	DHCPv4 Timeout
IPV4 STACK TURNED OFF	Phone is in IPv6-only mode with IPv4 Stack turned off
ILLEGAL IPV4 STATE	Illegal IPv4 state and should not happen

Table 44: DHCPv6 ethernet statistics messages

DHCPv6 State	Description
CDP INIT	CDP is initializing
DHCP6 BOUND	DHCPv6 is BOUND
DHCP6 DISABLED	DHCPv6 is DISABLED
DHCP6 RENEW	DHCPv6 is renewing
DHCP6 REBIND	DHCPv6 is rebinding
DHCP6 INIT	DHCPv6 is initializing
DHCP6 SOLICIT	DHCPv6 is soliciting
DHCP6 REQUEST	DHCPv6 is requesting

DHCPv6 State	Description
DHCP6 RELEASING	DHCPv6 is releasing
DHCP6 RELEASED	DHCPv6 is released
DHCP6 DISABLING	DHCPv6 is disabling
DHCP6 DECLINING	DHCPv6 is declining
DHCP6 DECLINED	DHCPv6 is declined
DHCP6 INFOREQ	DHCPv6 is INFOREQ
DHCP6 INFOREQ DONE	DHCPv6 is INFOREQ DONE
DHCP6 INVALID	DHCPv6 is INVALID; this is initial state
DISABLED DUPLICATE IPV6	DHCP6 is DISABLED, but DUPLICATE IPV6 DETECTED
DHCP6 DECLINED DUPLICATE IP	DHCP6 is DECLINED -- DUPLICATE IPV6 DETECTED
ROUTER ADVERTISE., (DUPLICATE IP)	Duplicated autoconfigured IPv6 address
DHCP6 WAITING COLDBOOT TIMEOUT	DHCPv6 is booting
DHCP6 TIMEOUT USING RESTORED VAL	DHCPv6 timeout, using the value saved in flash memory
DHCP6 TIMEOUT CANNOT RESTORE	DHCP6 timeout and there is no backup from flash memory
IPV6 STACK TURNED OFF	Phone is in IPv4-only mode with IPv6 Stack turned off
ROUTER ADVERTISE., (GOOD IP)	
ROUTER ADVERTISE., (BAD IP)	
UNRECOGNIZED MANAGED BY	IPv6 Address is not from router or DHCPv6 server
ILLEGAL IPV6 STATE	Illegal IPv6 state and should not happen

Display Wireless Statistics Screen

This procedure only applies to the wireless Cisco IP Phone 8861.

To display the Wireless Statistics screen, follow these steps:

Procedure

Step 1 Press **Applications** .

Step 2 Select **Admin settings**>**Status** > **Wireless Statistics**.

Step 3 To reset the Wireless statistics to 0, press **Clear**.

Step 4 To exit the Wireless Statistics screen, press **Exit**.

WLAN Statistics

The following table describes the WLAN statistics on the phone.

Table 45: WLAN Statistics on the Cisco Unified IP Phone

Item	Description
tx bytes	Number of bytes that the phone transmitted.
rx bytes	Number of bytes that the phone received.
tx packets	Number of packets that the phone transmitted.
rx packets	Number of packets that the phone received.
tx packets dropped	The number of packets dropped during transmission.
rx packets dropped	The number of packets dropped during reception.
tx packets errors	The number of erroneous packets that phone transmitted.
rx packets errors	The number of erroneous packets that phone received.
Tx frames	The number of successfully transmitted MSDU.
tx multicast frames	The nunber of successfully transmitted multicast MSDU.
tx retry	The number of MSDU that is successfully transmitted after one or more retransmissions.
tx multi retry	The number of multicast MSDU that is successfully transmitted after one or more retransmissions.
tx failure	This number of MSDU that is not transmitted successfully due to the number of transmit attempts exceeding the retry limit.
rts succeess	This counter shall increment when a CTS is received in response to an RTS.
rts failure	This counter shall increment when a CTS is not received in response to an RTS.
ack failure	This counter shall increment when an ACK is not received when expected.
rx duplicate frames	The number of received frame that the Sequence Control field indicates is a duplicate.
rx fraagmented packets	The number of successfully received MPDU of type Data or Management.
Roaming count	The number of succesful roaming.

Display Call Statistics Window

You can access the Call Statistics screen on the phone to display counters, statistics, and voice-quality metrics of the most recent call.

Note You can also remotely view the call statistics information by using a web browser to access the Streaming Statistics web page. This web page contains additional RTCP statistics that are not available on the phone.

A single call can use multiple voice streams, but data is captured for only the last voice stream. A voice stream is a packet stream between two endpoints. If one endpoint is put on hold, the voice stream stops even though the call is still connected. When the call resumes, a new voice packet stream begins, and the new call data overwrites the former call data.

Procedure

Step 1 Press **Applications** .

Step 2 Select **Admin settings** > **Status** > **Call statistics**.

Step 3 To exit the Call Statistics screen, press **Exit**.

Call Statistics Fields

The following table describes the items on the Call Statistics screen.

Table 46: Call Statistics items for the Cisco Unified Phone

Item	Description
Receiver codec	Type of received voice stream (RTP streaming audio from codec): • G.729 • G.722 • G722.2 AMR-WB • G.711 mu-law • G.711 A-law • iLBC • Opus • iSAC

Item	Description
Sender codec	Type of transmitted voice stream (RTP streaming audio from codec): • G.729 • G.722 • G722.2 AMR-WB • G.711 mu-law • G.711 A-law • iLBC. • Opus • iSAC
Receiver size	Size of voice packets, in milliseconds, in the receiving voice stream (RTP streaming audio).
Sender size	Size of voice packets, in milliseconds, in the transmitting voice stream.
Receiver packets	Number of RTP voice packets that were received since voice stream opened. **Note** This number is not necessarily identical to the number of RTP voice packets that were received since the call began because the call might have been placed on hold.
Sender packets	Number of RTP voice packets that were transmitted since voice stream opened. **Note** This number is not necessarily identical to the number of RTP voice packets that were transmitted since the call began because the call might have been placed on hold.
Avg jitter	Estimated average RTP packet jitter (dynamic delay that a packet encounters when going through the network), in milliseconds, that was observed since the receiving voice stream opened.
Max jitter	Maximum jitter, in milliseconds, that was observed since the receiving voice stream opened.
Receiver discarded	Number of RTP packets in the receiving voice stream that were discarded (bad packets, too late, and so on). **Note** The phone discards payload type 19 comfort noise packets that Cisco Gateways generate, because they increment this counter.
Receiver lost packets	Missing RTP packets (lost in transit).
Voice-Quality Metrics	

Item	Description
Cumulative conceal ratio	Total number of concealment frames divided by total number of speech frames that were received from start of the voice stream.
Interval conceal ratio	Ratio of concealment frames to speech frames in preceding 3-second interval of active speech. If using voice activity detection (VAD), a longer interval might be required to accumulate 3 seconds of active speech.
Max conceal ratio	Highest interval concealment ratio from start of the voice stream.
Conceal seconds	Number of seconds that have concealment events (lost frames) from the start of the voice stream (includes severely concealed seconds).
Severely conceal seconds	Number of seconds that have more than 5 percent concealment events (lost frames) from the start of the voice stream.
Latency	Estimate of the network latency, expressed in milliseconds. Represents a running average of the round-trip delay, measured when RTCP receiver report blocks are received.

Display Current Access Point Window

The Current Access Point screen displays statistics about the access point that the Cisco IP Phone 8861 uses for wireless communications.

Procedure

Step 1 Press **Applications** .

Step 2 Select **Admin settings** > **Status** > **Current Access Point**.

Step 3 To exit the Current Access Point screen, press **Exit**.

Current Access Points Fields

The following table describes the fields in the Current Access Point screen.

Table 47: Current Access Point Items

Item	Description
AP name	Name of the AP, if it is CCX-compliant; otherwise, the MAC address displays here.
MAC address	MAC address of the AP.
Frequency	The latest frequency where this AP was observed.
Current channel	The latest channel where this AP was observed.
Last RSSI	The latest RSSI in which this AP was observed.

Item	Description
Beacon interval	Number of time units between beacons. A time unit is 1.024 ms.
Capability	This field contains a number of subfields that are used to indicate requested or advertised optional capabilities.
Basic rates	Data rates that the AP requires and the AP at which the station must be capable of operating.
Optional rates	Data rates that the AP supports and the AP that are optional for the station to operate at.
Supported VHT(rx) rates	VHT Supported RX MCS Set received from AP.
Supported VHT(tx) rates	VHT Supported TX MCS Set received from AP.
Supported HT MCS	HT Supported MCS Set received from AP.
DTIM period	Every nth beacon is a dtime period. After each DTIM beacon, the AP sends any broadcast or multicast packets that are queued for power-save devices.
Country code	A two-digit country code. Country information might not be display if the country information element (IE) is not present in the beacon.
Channels	A list of supported channels (from the country IE).
Power constraint	The amount of power by which the maximum transmit power should be reduced from the regulatory domain limit.
Power limit	Maximum transmit power in dBm that is permitted for that channel.
Channel utilization	The percentage of time, normalized to 255, in which the AP sensed the medium was busy, as indicated by the physical or virtual carrier sense (CS) mechanism.
Station count	The total number of STAs currently associated with this AP.
Admission capacity	An unsigned integer that specifies the remaining amount of medium time that is available through explicit admission control, in units of 32 microseconds per second. If the value is 0, the AP does not support this information element and the capacity is unknown.
WMM supported	Support for Wi-Fi multimedia extensions.
UAPSD Supported	The AP supports Unscheduled Automatic Power Save Delivery. May only be available if WMM is supported. This feature is critical for talk time and for achieving maximum call density on the wireless IP Phone.
Proxy ARP	CCX-compliant AP supports responding to IP ARP requests on behalf of the associated station. This feature is critical to standby time on the wireless IP Phone.
CCX version	If the AP is CCX compliant, this field shows the CCX version.

Item	Description
Best Effort	Contains information related to the Best Effort queue.
Background	Contains information related to the Background queue.
Video	Contains information related to the Video queue.
Voice	Contains information related to the Voice queue.

Cisco IP Phone Web Page

Each Cisco IP Phone has a web page from which you can view a variety of information about the phone, including:

- Device information: Displays device settings and related information for the phone.
- Network setup: Displays network setup information and information about other phone settings.
- Network statistics: Displays hyperlinks that provide information about network traffic.
- Device logs: Displays hyperlinks that provide information that you can use for troubleshooting.
- Streaming statistic: Displays hyperlinks that display a variety of streaming statistics.
- System: Displays a hyperlink to restart the phone.

This section describes the information that you can obtain from the phone web page. You can use this information to remotely monitor the operation of a phone and to assist with troubleshooting.

You can also obtain much of this information directly from a phone.

Access Web Page for Phone

To access the web page for a phone, follow these steps:

Note If you cannot access the web page, it may be disabled by default.

Procedure

Step 1 Obtain the IP address of the Cisco IP Phone by using one of these methods:

a) Search for the phone in Cisco Unified Communications Manager Administration by choosing **Device** > **Phone**. Phones that register with Cisco Unified Communications Manager display the IP address on the **Find and List Phones** window and at the top of the **Phone Configuration** window.

b) On the Cisco IP Phone, press **Applications**, choose **Admin settings** > **Network setup** > **Ethernet setup** > **IPv4 setup**, and then scroll to the IP Address field.

Step 2 Open a web browser and enter the following URL, where *IP_address* is the IP address of the Cisco IP Phone:

`http://`*IP_address*

Device Information

The Device information area on a phone web page displays device settings and related information for the phone. The following table describes these items.

Note Some of the items in the following table do not apply to all phone models.

To display the **Device information** area, access the web page for the phone as described in Access Web Page for Phone, on page 222, and then click the **Device information** hyperlink.

Table 48: Device Information Area Items

Item	Description
Service mode	The service mode for the phone.
Service name	The domain for the service.
Service state	The current state of the service.
MAC Address	Media Access Control (MAC) address of the phone.
Host name	Unique, fixed name that is automatically assigned to the phone based on the MAC address.
Phone DN	Directory number that is assigned to the phone.
App load ID	Application firmware version that is running on the phone.
Boot load ID	Boot firmware version.
Version	Identifier of the firmware that is running on the phone.
Key expansion module 1	Identifier for the first key expansion module, if applicable. Applicable to Cisco IP Phone 8851, 8851NR, 8861, 8865, and 8865NR.
Key expansion module 2	Identifier for the second key expansion module, if applicable. Applicable to Cisco IP Phone 8851, 8851NR, 8861, 8865, and 8865NR.
Key expansion module 3	Identifier for the third key expansion module, if applicable. Applicable to Cisco IP Phone 8851, 8851NR, 8861, 8865, and 8865NR.
Hardware revision	Minor revision value of the phone hardware.
Serial number	Unique serial number of the phone.
Model number	Model number of the phone.

Item	Description
Message waiting	Indicates whether is a voice message is waiting on the primary line for this phone.
UDI	Displays the following Cisco Unique Device Identifier (UDI) information about the phone: • Device type—Indicates hardware type. For example, phone displays for all phone models. • Device description—Displays the name of the phone associated with the indicated model type. • Product identifier—Specifies the phone model. • Version ID (VID)—Specifies the major hardware version number. • Serial number—Displays the unique serial number of the phone.
Key expansion module UDI	Cisco Unique Device Identifier (UDI) of the key expansion module. Applicable to Cisco IP Phone 8851, 8851NR, 8861, 8865, and 8865NR.
Name of the Headset	Displays the name of the attached Cisco headset in the left column. The right column contains this information: • Port—Displays how the headset connects to the phone. • USB • AUX • Version—Displays the headset firmware version. • Radio range—Displays the strength configured for the DECT radio. Applicable to the Cisco Headset 560 Series only. • Bandwidth—Displays if the headset uses Wide band or Narrow band. Applicable to the Cisco Headset 560 Series only. • Bluetooth—Displays if Bluetooth is enabled or disabled. Applicable to the Cisco Headset 560 Series only. • Conference—Displays if the conference feature is enabled or disabled. Applicable to the Cisco Headset 560 Series only. • Firmware source—Displays the permitted firmware upgrade method: • Restrict to UCM only • Allow from UCM or Cisco Cloud Applicable to the Cisco Headset 560 Series only.
Time	Time for the Date/Time Group to which the phone belongs. This information comes from Cisco Unified Communications Manager.

Item	Description
Time zone	Time zone for the Date/Time Group to which the phone belongs. This information comes from Cisco Unified Communications Manager.
Date	Date for the Date/Time Group to which the phone belongs. This information comes from Cisco Unified Communications Manager.
System free memory	Amount of unused memory on the phone
Java heap free memory	Amount of free internal Java heap memory
Java pool free memory	Amount of free internal Java pool memory
FIPS mode enabled	Indicates if the Federal Information Processing Standard (FIPS) Mode is enabled.

Network Setup

The Network setup area on a phone web page displays network setup information and information about other phone settings. The following table describes these items.

You can view and set many of these items from the Network Setup menu on the Cisco IP Phone.

Note Some of the items in the following table do not apply to all phone models.

To display the **Network Setup** area, access the web page for the phone as described in Access Web Page for Phone, on page 222, and then click the **Network setup** hyperlink.

Table 49: Network Setup Area Items

Item	Description
MAC address	Media Access Control (MAC) address of the phone.
Host name	Host name that the DHCP server assigned to the phone.
Domain name	Name of the Domain Name System (DNS) domain in which the phone resides.
DHCP server	IP address of the Dynamic Host Configuration Protocol (DHCP) server from which the pl the IP address.
BOOTP server	Indicates whether the phone obtains the configuration from a Bootstrap Protocol (BootP)
DHCP	Indicates whether the phone uses DHCP.
IP address	Internet Protocol (IPv4) address of the phone.
Subnet mask	Subnet mask that the phone uses.
Default router	Default router used that the phone uses.

Item	Description
DNS server 1–3	Primary Domain Name System (DNS) server (DNS Server 1) and optional backup DNS serve Server 2 and 3) that the phone uses.
Alternate TFTP	Indicates whether the phone is using an alternative TFTP server.
TFTP server 1	Primary Trivial File Transfer Protocol (TFTP) server used that the phone uses.
TFTP server 2	Backup Trivial File Transfer Protocol (TFTP) server used that the phone uses.
DHCP address released	Indicates the setting of the DHCP address rReleased option on the phone Network Configurati
Operational VLAN ID	Operational Virtual Local Area Network (VLAN) that is configured on a Cisco Catalyst swit which the phone is a member.
Admin VLAN ID	Auxiliary VLAN in which the phone is a member.
CUCM server1–5	Host names or IP addresses, in prioritized order, of the Cisco Unified Communications Manage with which the phone can register. An item can also show the IP address of an SRST router t capable of providing limited Cisco Unified Communications Manager functionality, if such a available. For an available server, an item shows the Cisco Unified Communications Manager server IP and one of the following states: • Active—Cisco Unified Communications Manager server from which the phone is curre receiving call-processing services • Standby—Cisco Unified Communications Manager server to which the phone switches current server becomes unavailable • Blank—No current connection to this Cisco Unified Communications Manager server An item may also include the Survivable Remote Site Telephony (SRST) designation, which i an SRST router capable of providing Cisco Unified Communications Manager functionality limited feature set. This router assumes control of call processing if all other Cisco Unified Communications Manager servers become unreachable. The SRST Cisco Unified Communi Manager always appears last in the list of servers, even if it is active. You configure the SRS address in the Device Pool section in Cisco Unified Communications Manager Configuration
Information URL	URL of the help text that appears on the phone.
Directories URL	URL of the server from which the phone obtains directory information.
Messages URL	URL of the server from which the phone obtains message services.
Services URL	URL of the server from which the phone obtains Cisco Unified IP Phone services.
Idle URL	URL that the phone displays when the phone is idle for the time that the Idle URL Time field and no menu is open.
Idle URL time	Number of seconds that the phone is idle and no menu is open before the XML service that t URL specifies activates.
Proxy server URL	URL of proxy server, which makes HTTP requests to nonlocal host addresses on behalf of th HTTP client and provides responses from the nonlocal host to the phone HTTP client.

Item	Description
Authentication URL	URL that the phone uses to validate requests that are made to the phone web server.
SW port setup	Speed and duplex of the switch port, where: • A = Auto Negotiate • 10H = 10-BaseT/half duplex • 10F = 10-BaseT/full duplex • 100H = 100-BaseT/half duplex • 100F = 100-BaseT/full duplex • 1000F = 1000-BaseT/full duplex • No Link= No connection to the switch port
PC port setup	Speed and duplex of the PC port, where: • A = Auto Negotiate • 10H = 10-BaseT/half duplex • 10F = 10-BaseT/full duplex • 100H = 100-BaseT/half duplex • 100F = 100-BaseT/full duplex • 1000F = 1000-BaseT/full duplex • No Link = No connection to the PC port
PC port disabled	Indicates whether the PC port on the phone is enabled or disabled.
User locale	User locale that associates with the phone user. Identifies a set of detailed information to su including language, font, date and time formatting, and alphanumeric keyboard text infor
Network locale	Network locale that associates with the phone user. Identifies a set of detailed information the phone in a specific location, including definitions of the tones and cadences that the p
User locale version	Version of the user locale that is loaded on the phone.
Network locale version	Version of the network locale that is loaded on the phone.
Speaker enabled	Indicates whether the speakerphone is enabled on the phone.
GARP enabled	Indicates whether the phone learns MAC addresses from Gratuitous ARP responses.
Span to PC port	Indicates whether the phone forwards packets that are transmitted and received on the net the access port.
Video capability enabled	Indicates whether the phone can participate in video calls when it connects to an appropriat camera.
Voice VLAN enabled	Indicates whether the phone allows a device that is attached to the PC port to access the V
PC VLAN enabled	VLAN that identifies and removes 802.1P/Q tags from packets that are sent to the PC.
Auto line select enabled	Identifies if the phone automatically selects a line when the phone goes off hook.
DSCP protocol control	DSCP IP classification for call control signaling.

Item	Description
DSCP for configuration	DSCP IP classification for any phone configuration transfer.
DSCP for services	DSCP IP classification for phone-based services.
Security mode (nonsecure)	Security mode that is set for the phone.
Web access enabled	Indicates whether web access is enabled (Yes) or disabled (No) for the phone.
SSH access enabled	Indicates if the SSH port has been enabled or disabled.
CDP: SW Port	Indicates whether CDP support exists on the switch port (default is enabled). Enable CDP on the switch port for VLAN assignment for the phone, power negotiation, QoS management, and 802.1x security. Enable CDP on the switch port when the phone connects to a Cisco switch. When CDP is disabled in Cisco Unified Communications Manager, a warning is presented, in that CDP should be disabled on the switch port only if the phone connects to a non-Cisco sw The current PC and switch port CDP values are shown on the Settings menu.
CDP: PC Port	Indicates whether CDP is supported on the PC port (default is enabled). When CDP is disabled in Cisco Unified Communications Manager, a warning is displays to i that disabling CDP on the PC port prevents CVTA from working. The current PC and switch port CDP values are shown in the Settings menu.
LLDP-MED:SW Port	Indicates whether Link Layer Discovery Protocol Media Endpoint Discovery (LLDP-MED) is on the switch port.
LLDP-MED:PC Port	Indicates whether LLDP-MED is enabled on the PC port.
LLDP Power Priority	Phone power priority to the switch, thus enabling the switch to appropriately provide power t phones. Settings include: • Unknown: This is the default value. • Low • High • Critical
LLDP Asset ID	Asset ID that is assigned to the phone for inventory management.
CTL file	MD5 hash of the CTL file.
ITL file	The ITL file contains the initial trust list.
ITL signature	MD5 hash of the ITL file
CAPF server	CPF server in use
TVS	The main component of Security by Default. Trust Verification Services (TVS) enables Cisco IP Phones to authenticate application servers, such as EM services, directory, and MIDlet, du HTTPS establishment.

Item	Description
TFTP server	The name of the TFTP Server used by the phone.
TFTP server	The name of the TFTP Server used by the phone.
Automatic port synchronization	Indicates if the phone automatically synchronizes port speed to eliminate packet loss.
Switch port remote configuration	Indicates if the SW port is remotely controlled.
PC port remote configuration	Indicates if the PC port is remotely controlled.
IP addressing mode	Identifies the addressing mode: • IPv4 Only • IPv4 and IPv6 • IPv6 Only
IP preference mode control	Indicates the IP address version that the phone uses during signaling with Cisco Unified Com Manager when both IPv4 and IPv6 are both available on the phone.
IP preference mode for media	
IPv6 auto configuration	Indicates that for media the device uses an IPv4 address to connect to the Cisco Unified Com Manager.
IPv6 duplicate address protection	
IPv6 accept redirect message	Indicates if phone accepts the redirect messages from the same router that is used for the number.
IPv6 reply multicast echo request	Indicates that the phone sends an Echo Reply message in response to an Echo Request m to an IPv6-only address.
IPv6 load server	Used to optimize installation time for phone firmware upgrades and off load the WAN by images locally, negating the need to traverse the WAN link for each phone's upgrade.
IPv6 log server	
IPv6 CAPF server	Indicates the IP address and port of the remote logging machine to which the phone sends lo
DHCPv6	Indicates the method that the phone uses to get the IPv6-only address. When DHCPv6 is enabled, the phone gets the IPv6 address either from DHCPv6 server or fi by RA sent by the IPv6-enabled router. And if DHCPv6 is disabled, the phone will not have (from DHCPv6 server) or stateless (from SLAAC) IPv6 address. **Note** Unlike DHCPv4, even DHCPv6 is disabled the phone can still generate a SLA if autoconfigure is enabled.

Item	Description
IPv6 address	Displays the current IPv6-only address of the phone. Two address formats are supported: • Eight sets of hexadecimal digits separated by colons X:X:X:X:X:X:X:X • Compressed format to collapse a single run of consecutive zero groups into a single gro represented by a double colon.
IPv6 prefix length	Displays the current IPv6-only prefix length for the subnet.
IPv6 default router	Displays the default IPv6 router used by the phone.
IPv6 DNS server 1–2	Displays the primary and secondary DNSv6 server used by the phone
IPv6 Alternate TFTP	Displays if an alternate IPv6 TFTP server is used.
IPv6 TFTP server 1–2	Displays the primary and secondary IPv6 TFTP server used by the phone.
IPv6 address released	Displays if the user has released the IPv6-related information.
EnergyWise power level	The power level that is used when the phone is sleeping.
EnergyWise domain	The EnergyWise domain that the phone is in.
DF_BIT	Indicates the DF bit setting for packets.

Network Statistics

The following Network statistics hyperlinks on a phone web page provide information about network traffic on the phone:

- Ethernet information: Displays information about Ethernet traffic.
- Access: Displays information about network traffic to and from the PC port on the phone.
- Network: Displays information about network traffic to and from the network port on the phone.

To display a Network statistics area, access the web page for the phone, and then click the **Ethernet Information**, the **Access**, or the **Network** hyperlink.

Ethernet Information Web Page

The following table describes the contents of the Ethernet information web page.

Table 50: Ethernet Information Items

Item	Description
Tx Frames	Total number of packets that the phone transmits.
Tx broadcast	Total number of broadcast packets that the phone transmits.
Tx multicast	Total number of multicast packets that the phone transmits.

Item	Description
Tx unicast	Total number of unicast packets that the phone transmits.
Rx Frames	Total number of packets received by the phone.
Rx broadcast	Total number of broadcast packets that the phone receives..
Rx multicast	Total number of multicast packets that the phone receives.
Rx unicast	Total number of unicast packets that the phone receives.
Rx PacketNoDes	Total number of shed packets that the no Direct Memory Access (DMA) descriptor causes.

Access and Network Web Pages

The following table describes the information in the Access and Network web pages.

Table 51: Access and Network Fields

Item	Description
Rx totalPkt	Total number of packets that the phone received.
Rx crcErr	Total number of packets that were received with CRC failed.
Rx alignErr	Total number of packets between 64 and 1522 bytes in length that were received and that have a bad Frame Check Sequence (FCS).
Rx multicast	Total number of multicast packets that the phone received.
Rx broadcast	Total number of broadcast packets that the phone received.
Rx unicast	Total number of unicast packets that the phone received.
Rx shortErr	Total number of received FCS error packets or Align error packets that are less than 64 bytes in size.
Rx shortGood	Total number of received good packets that are less than 64 bytes size.
Rx longGood	Total number of received good packets that are greater than 1522 bytes in size.
Rx longErr	Total number of received FCS error packets or Align error packets that are greater than 1522 bytes in size.
Rx size64	Total number of received packets, including bad packets, that are between 0 and 64 bytes in size.
Rx size65to127	Total number of received packets, including bad packets, that are between 65 and 127 bytes in size.
Rx size128to255	Total number of received packets, including bad packets, that are between 128 and 255 bytes in size.

Item	Description
Rx size256to511	Total number of received packets, including bad packets, that are between 256 and 511 bytes in size.
Rx size512to1023	Total number of received packets, including bad packets, that are between 512 and 1023 bytes in size.
Rx size1024to1518	Total number of received packets, including bad packets, that are between 1024 and 1518 bytes in size.
Rx tokenDrop	Total number of packets that were dropped due to lack of resources (for example, FIFO overflow).
Tx excessDefer	Total number of packets that were delayed from transmitting due to busy medium.
Tx lateCollision	Number of times that collisions occurred later than 512 bit times after the start of packet transmission.
Tx totalGoodPkt	Total number of good packets (multicast, broadcast, and unicast) that the phone received.
Tx Collisions	Total number of collisions that occurred while a packet was transmitted.
Tx excessLength	Total number of packets that were not transmitted because the packet experienced 16 transmission attempts.
Tx broadcast	Total number of broadcast packets that the phone transmitted.
Tx multicast	Total number of multicast packets that the phone transmitted.
LLDP FramesOutTotal	Total number of LLDP frames that the phone sent out.
LLDP AgeoutsTotal	Total number of LLDP frames that timed out in the cache.
LLDP FramesDiscardedTotal	Total number of LLDP frames that were discarded when any of the mandatory TLVs is missing, out of order, or contains out of range string length.
LLDP FramesInErrorsTotal	Total number of LLDP frames that were received with one or more detectable errors.
LLDP FramesInTotal	Total number of LLDP frames that the phone receives.
LLDP TLVDiscardedTotal	Total number of LLDP TLVs that are discarded.
LLDP TLVUnrecognizedTotal	Total number of LLDP TLVs that are not recognized on the phone.
CDP Neighbor Device ID	Identifier of a device connected to this port that CDP discovered.
CDP Neighbor IPv6 address	IP address of the neighbor device discovered that CDP protocol discovered.
CDP Neighbor Port	Neighbor device port to which the phone is connected discovered by CDP protocol.

Item	Description
LLDP Neighbor Device ID	Identifier of a device connected to this port discovered by LLDP discovered.
LLDP Neighbor IPv6 address	IP address of the neighbor device that LLDP protocol discovered.
LLDP Neighbor Port	Neighbor device port to which the phone connects that LLDP protocol discovered.
Port Information	Speed and duplex information.

Device Logs

The following Device log hyperlinks on a phone web page provide information that helps to monitor and troubleshoot the phone.

- Console logs: Includes hyperlinks to individual log files. The console log files include debug and error messages that the phone received.
- Core dumps: Includes hyperlinks to individual dump files. The core dump files include data from a phone crash.
- Status messages: Displays the 10 most recent status messages that the phone has generated since it last powered up. The Status Messages screen on the phone also displays this information.
- Debug display: Displays debug messages that might be useful to Cisco TAC if you require assistance with troubleshooting.

Streaming Statistics

A Cisco Unified IP Phone can stream information to and from up to three devices simultaneously. A phone streams information when it is on a call or is running a service that sends or receives audio or data.

The Streaming statistics areas on a phone web page provide information about the streams.

The following table describes the items in the Streaming Statistics areas.

Table 52: Streaming Statistics area items

Item	Description
Remote address	IP address and UDP port of the destination of the stream.
Local address	IP address and UPD port of the phone.
Start time	Internal time stamp indicates when Cisco Unified Communications Manager request phone start transmitting packets.
Stream status	Indication of whether streaming is active or not.
Host name	Unique, fixed name that is automatically assigned to the phone based on the MAC a
Sender packets	Total number of RTP data packets that the phone transmitted since it started this co The value is 0 if the connection is set to receive-only mode.

Item	Description
Sender octets	Total number of payload octets that the phone transmitted in RTP data packets since it s this connection. The value is 0 if the connection is set to receive-only mode.
Sender codec	Type of audio encoding that is for the transmitted stream.
Sender reports sent (see note)	Number of times the RTCP Sender Report has been sent.
Sender report time sent (see note)	Internal time-stamp indication as to when the last RTCP Sender Report was sent.
Rcvr lost packets	Total number of RTP data packets that have been lost since data reception started on th connection. Defined as the number of expected packets less the number of packets actu received, where the number of received packets includes any that are late or are duplicat value displays as 0 if the connection was set to send-only mode.
Avg jitter	Estimate of mean deviation of the RTP data packet interarrival time, measured in millise The value displays as 0 if the connection was set to send-only mode.
Receiver codec	Type of audio encoding that is used for the received stream.
Receiver reports sent (see note)	Number of times the RTCP Receiver Reports have been sent.
Receiver report time sent (see note)	Internal time-stamp indication as to when a RTCP Receiver Report was sent.
Rcvr packets	Total number of RTP data packets that the phone has received since data reception start this connection. Includes packets that were received from different sources if this call i multicast call. The value displays as 0 if the connection was set to send-only mode.
Rcvr octets	Total number of payload octets that the device received in RTP data packets since recep started on the connection. Includes packets that were received from different sources if t is a multicast call. The value displays as 0 if the connection was set to send-only mode.
MOS LQK	Score that is an objective estimate of the mean opinion score (MOS) for listening quality that rates from 5 (excellent) to 1(bad). This score is based on audible concealment ever are to frame loss in the preceding eight-second interval of the voice stream. For more infor see Voice Quality Monitoring, on page 262. **Note** The MOS LQK score can vary due to the codec type that the Cisco Unifie Phone uses.
Avg MOS LQK	Average MOS LQK score that was observed for the entire voice stream.
Min MOS LQK	Lowest MOS LQK score that was observed from the start of the voice stream.

Item	Description
Max MOS LQK	Baseline or highest MOS LQK score that was observed from start of the voice strea These codecs provide the following maximum MOS LQK score under normal cond no frame loss: • G.711 yields 4.5. • G.729 A /AB yields 3.7.
MOS LQK version	Version of the Cisco proprietary algorithm that is used to calculate MOS LQK score
Cumulative conceal ratio	Total number of concealment frames divided by total number of speech frames that we from the start of the voice stream.
Interval conceal ratio	Ratio of concealment frames to speech frames in the preceding 3-second interval of speech. If voice activity detection (VAD) is in use, a longer interval might be requir accumulate three seconds of active speech.
Max conceal ratio	Highest interval concealment ratio from the start of the voice stream.
Conceal secs	Number of seconds that have concealment events (lost frames) from the start of the vo (includes severely concealed seconds).
Severely conceal secs	Number of seconds that have more than five percent concealment events (lost frames start of the voice stream.
Latency (see note)	Estimate of the network latency, expressed in milliseconds. Represents a running av the round-trip delay, measured when RTCP receiver report blocks are received.
Max jitter	Maximum value of instantaneous jitter, in milliseconds.
Sender size	RTP packet size, in milliseconds, for the transmitted stream.
Sender reports received (see note)	Number of times RTCP Sender Reports have been received.
Sender report time received (see note)	Most recent time when an RTCP Sender Report was received.
Receiver size	RTP packet size, in milliseconds, for the received stream.
Receiver discarded	RTP packets that were received from the network but were discarded from the jitter
Receiver reports received (see note)	Number of times RTCP Receiver Reports have been received.
Receiver report time received (see note)	Most recent time when an RTCP Receiver Report was received.
Rcvr encrypted	Indicates if the receiver is using encryption.
Sender encrypted	Indicates if the sender is using encryption.

Item	Description
Sender frames	Number of frames sent.
Sender partial frames	Number of partial frames sent.
Sender i frames	Number of I frames sent. I frames are used in video transmission.
Sender IDR frames	Number of instantaneous decoder refresh (IDR) frames sent. IDR frames are used in vi transmission.
Sender frame rate	Rate at which the sender is sending frames.
Sender bandwidth	Bandwidth for the sender.
Sender resolution	Video resolution of the sender.
Rcvr frames	Number of frames received
Rcvr partial frames	Number of partial frames received
Rcvr i frames	Number of I frames received.
Rcvr IDR frames	Number of IDR frames received.
Rcvr IFrames req	Number of requested IDR frames received
Rcvr frame rate	Rate at which the receiver is receiving frames.
Rcvr frames lost	Number of frames that were not received.
Rcvr frame errors	Number of frames that were not received.
Rcvr bandwidth	Bandwidth of the receiver.
Rcvr resolution	Video resolution of the receiver.
Domain	Domain that the phone resides in.
Sender joins	Number of times the sender joined.
Rcvr joins	Number of times the receiver joined
Byes	Number of “Bye” frames
Sender start time	Time that the sender started.
Rcvr start time	Time that the receiver started.
Row status	Whether the phone is streaming
Sender tool	Type of audio encoding used for the stream
Sender reports	RTCP Sender Reports
Sender report time	Last time at which an RTCP Sender Report was sent.

Item	Description
Rcvr Jitter	Maximum jitter of stream
Receiver tool	Type of audio encoding used for the stream
Rcvr reports	Number of times this streaming statistics report has been accessed from the web pa
Rcvr report time	Internal time stamp indicating when this streaming statistics report was generated
Is video	Indicates if the call was a video call or audio only.
Call ID	Identification of the call
Group ID	Identification of the group that the phone is in.

Note When the RTP Control Protocol is disabled, no data generates for this field and thus displays as 0.

Request Information from the Phone in XML

For troubleshooting purposes, you can request information from the phone. The resulting information is in XML format. The following information is available:

- CallInfo is call session information for a specific line.
- LineInfo is line configuration information for the phone.
- ModeInfo is phone mode information.

Before you begin

Web access needs to be enabled to get the information.

The phone must be associated with a user.

Procedure

Step 1 For Call Info, enter the following URL in a browser: **http://<phone ip address>/CGI/Java/CallInfo<x>**

where

- <*phone ip address*> is the IP address of the phone
- <*x*> is the line number to obtain information about.

The command returns an XML document.

Step 2 For Line Info, enter the following URL in a browser: **http://<phone ip address>/CGI/Java/LineInfo**

where

- <*phone ip address*> is the IP address of the phone

The command returns an XML document.

Step 3 For Model Info, enter the following URL in a browser: **http://<phone ip address>/CGI/Java/ModeInfo**

where

- <*phone ip address*> is the IP address of the phone

The command returns an XML document.

Sample CallInfo Output

The following XML code is an example of the output from the CallInfo command.

```
<?xml version="1.0" encoding="UTF-8"?>
<CiscoIPPhoneCallLineInfo>
  <Prompt/>
  <Notify/>
  <Status/>
  <LineDirNum>1030</LineDirNum>
  <LineState>CONNECTED</LineState>
  <CiscoIPPhoneCallInfo>
     <CallState>CONNECTED</CallState>
     <CallType>INBOUND</CallType>
     <CallingPartyName/>
     <CallingPartyDirNum>9700</CallingPartyDirNum>
     <CalledPartyName/>
     <CalledPartyDirNum>1030</CalledPartyDirNum>
     <HuntPilotName/>
     <CallReference>30303060</CallReference>
     <CallDuration>12835</CallDuration>
     <CallStatus>null</CallStatus>
     <CallSecurity>UNAUTHENTICATED</CallSecurity>
     <CallPrecedence>ROUTINE</CallPrecedence>
     <FeatureList/>
   </CiscoIPPhoneCallInfo>
   <VisibleFeatureList>
     <Feature Position="1" Enabled="true" Label="End Call"/>
     <Feature Position="2" Enabled="true" Label="Show Detail"/>
   </VisibleFeatureList>
</CiscoIPPhoneCallLineInfo>
```

Sample LineInfo Output

The following XML code is an example of the output from the LineInfo command.

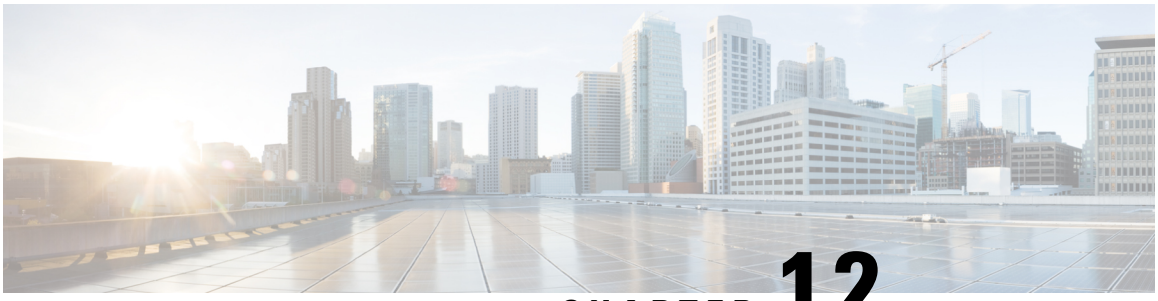
```
<CiscoIPPhoneLineInfo>
   <Prompt/>
   <Notify/>
   <Status>null</Status>
   <CiscoIPPhoneLines>
     <LineType>9</LineType>
     <lineDirNum>1028</lineDirNum>
```

```
      <MessageWaiting>NO</MessageWaiting>
      <RingerName>Chirp1</RingerName>
      <LineLabel/>
      <LineIconState>ONHOOK</LineIconState>
    </CiscoIPPhoneLines>
    <CiscoIPPhoneLines>
      <LineType>9</LineType>
      <lineDirNum>1029</lineDirNum>
      <MessageWaiting>NO</MessageWaiting>  <RingerName>Chirp1</RingerName>
      <LineLabel/>
      <LineIconState>ONHOOK</LineIconState>
    </CiscoIPPhoneLines>
    <CiscoIPPhoneLines>
      <LineType>9</LineType>
      <lineDirNum>1030</lineDirNum>
      <MessageWaiting>NO</MessageWaiting>
      <RingerName>Chirp1</RingerName>
      <LineLabel/>
      <LineIconState>CONNECTED</LineIconState>
    </CiscoIPPhoneLines>
    <CiscoIPPhoneLines>
      <LineType>2</LineType>
      <lineDirNum>9700</lineDirNum>
      <MessageWaiting>NO</MessageWaiting>
      <LineLabel>SD9700</LineLabel>
      <LineIconState>ON</LineIconState>
  </CiscoIPPhoneLines>
</CiscoIPPhoneLineInfo>
```

Sample ModeInfo Output

The following XML code is an example of the output from the ModeInfo command.

```
<?xml version="1.0" encoding="utf-8"?>
<CiscoIPPhoneModeInfo>
   <PlaneTitle>Applications</PlaneTitle>
   <PlaneFieldCount>12</PlaneFieldCount>
   <PlaneSoftKeyIndex>0</PlaneSoftKeyIndex>
   <PlaneSoftKeyMask>0</PlaneSoftKeyMask>
   <Prompt></Prompt>
   <Notify></Notify>
   <Status></Status>
   <CiscoIPPhoneFields>
      <FieldType>0</FieldType>
      <FieldAttr></FieldAttr>
      <fieldHelpIndex>0</fieldHelpIndex>
      <FieldName>Call History</FieldName>
      <FieldValue></FieldValue>
   </CiscoIPPhoneFields>
   <CiscoIPPhoneFields>
      <FieldType>0</FieldType>
      <FieldAttr></FieldAttr>
      <fieldHelpIndex>0</fieldHelpIndex>
      <FieldName>Preferences</FieldName>
      <FieldValue></FieldValue>
   </CiscoIPPhoneFields>
   ...
</CiscoIPPhoneModeInfo>
```



CHAPTER 12

Troubleshooting

- General Troubleshooting Information, on page 241
- Startup Problems, on page 242
- Phone Reset Problems, on page 246
- Phone Cannot Connect to LAN, on page 248
- Cisco IP Phone Security Problems, on page 249
- Video Call Problems, on page 251
- General Telephone Call Problems, on page 252
- Troubleshooting Procedures, on page 253
- Control Debug Information from Cisco Unified Communications Manager, on page 257
- Additional Troubleshooting Information, on page 258

General Troubleshooting Information

The following table provides general troubleshooting information for the Cisco IP Phone.

Table 53: Cisco IP Phone troubleshooting

Summary	Explanation
Connecting a Cisco IP Phone to another Cisco IP Phone	Cisco does not support connecting an IP phone to another IP Phone thr port. Each IP Phone should connect directly to a switch port. If phones a together in a line by using the PC port, the phones do not work.
Prolonged broadcast storms cause IP phones to reset, or be unable to make or answer a call	A prolonged Layer 2 broadcast storm (lasting several minutes) on the v may cause IP phones to reset, lose an active call, or be unable to initiat call. Phones may not come up until a broadcast storm ends.

Summary	Explanation
Moving a network connection from the phone to a workstation	If you power your phone through the network connection, you must be care decide to unplug the network connection of the phone and plug the cable into computer. **Caution** The network card in the computer cannot receive power throu network connection; if power comes through the connection, the card can be destroyed. To protect a network card, wait 10 secc longer after unplugging the cable from the phone before pluggi a computer. This delay gives the switch enough time to recogr there is no longer a phone on the line and to stop providing po the cable.
Changing the telephone configuration	By default, the network configuration options are locked to prevent users fror changes that could impact their network connectivity. You must unlock the configuration options before you can configure them. See Apply a Phone P on page 48 for details. **Note** If the administrator password is not set in common phone prof user can modify the network settings.
Codec mismatch between the phone and another device	The RxType and the TxType statistics show the codec that is used for a con between this Cisco IP Phone and the other device. The values of these statisti match. If they do not, verify that the other device can handle the codec conv or that a transcoder is in place to handle the service.
Sound sample mismatch between the phone and another device	The RxSize and the TxSize statistics show the size of the voice packets that in a conversation between this Cisco IP Phone and the other device. The va these statistics should match.
Loopback condition	A loopback condition can occur when the following conditions are met: • The SW Port Configuration option in the Network Configuration men phone is set to 10 Half (10-BaseT/half duplex). • The phone receives power from an external power supply. • The phone is powered down (the power supply is disconnected). In this case, the switch port on the phone can become disabled and the foll message appears in the switch console log: HALF_DUX_COLLISION_EXCEED_THRESHOLD To resolve this problem, reenable the port from the switch.

Startup Problems

After you install a phone into your network and add it to Cisco Unified Communications Manager, the phone should start up as described in the related topic below.

If the phone does not start up properly, see the following sections for troubleshooting information.

Related Topics

Phone Startup Verification, on page 62

Cisco IP Phone Does Not Go Through the Normal Startup Process

Problem

When you connect a Cisco IP Phone to the network port, the phone does not go through the normal startup process as described in the related topic and the phone screen does not display information.

Cause

If the phone does not go through the startup process, the cause may be faulty cables, bad connections, network outages, lack of power, or the phone may not be functional.

Solution

To determine whether the phone is functional, use the following suggestions to eliminate other potential problems.

- Verify that the network port is functional:
  - Exchange the Ethernet cables with cables that you know are functional.
  - Disconnect a functioning Cisco IP Phone from another port and connect it to this network port to verify that the port is active.
  - Connect the Cisco IP Phone that does not start up to a different network port that is known to be good.
  - Connect the Cisco IP Phone that does not start up directly to the port on the switch, eliminating the patch panel connection in the office.
- Verify that the phone is receiving power:
  - If you are using external power, verify that the electrical outlet is functional.
  - If you are using in-line power, use the external power supply instead.
  - If you are using the external power supply, switch with a unit that you know to be functional.
- If the phone still does not start up properly, power up the phone from the backup software image.
- If the phone still does not start up properly, perform a factory reset of the phone.
- After you attempt these solutions, if the phone screen on the Cisco IP Phone does not display any characters after at least five minutes, contact a Cisco technical support representative for additional assistance.

Related Topics

Phone Startup Verification, on page 62

Cisco IP Phone Does Not Register with Cisco Unified Communications Manager

If the phone proceeds past the first stage of the startup process (LED buttons flashing on and off) but continues to cycle through the messages that displays on the phone screen, the phone is not starting up properly. The phone cannot successfully start up unless it connects to the Ethernet network and it registers with a Cisco Unified Communications Manager server.

In addition, problems with security may prevent the phone from starting up properly. See Troubleshooting Procedures, on page 253 for more information.

Phone Displays Error Messages

Problem

Status messages display errors during startup.

Solution

While the phone cycles through the startup process, you can access status messages that might provide you with information about the cause of a problem.

Related Topics

Display Status Messages Window, on page 209

Phone Cannot Connect to TFTP Server or to Cisco Unified Communications Manager

Problem

If the network is down between the phone and either the TFTP server or Cisco Unified Communications Manager, the phone cannot start up properly.

Solution

Ensure that the network is currently running.

Phone Cannot Connect to TFTP Server

Problem

The TFTP server settings may not be correct.

Solution

Check the TFTP settings.

Related Topics

Check TFTP Settings, on page 254

Phone Cannot Connect to Server

Problem

The IP addressing and routing fields may not be configured correctly.

Solution

You should verify the IP addressing and routing settings on the phone. If you are using DHCP, the DHCP server should provide these values. If you have assigned a static IP address to the phone, you must enter these values manually.

Phone Cannot Connect Using DNS

Problem

The DNS settings may be incorrect.

Solution

If you use DNS to access the TFTP server or Cisco Unified Communications Manager, you must ensure that you specify a DNS server.

Cisco Unified Communications Manager and TFTP Services Are Not Running

Problem

If the Cisco Unified Communications Manager or TFTP services are not running, phones may not be able to start up properly. In such a situation, it is likely that you are experiencing a systemwide failure, and other phones and devices are unable to start up properly.

Solution

If the Cisco Unified Communications Manager service is not running, all devices on the network that rely on it to make phone calls are affected. If the TFTP service is not running, many devices cannot start up successfully. For more information, see Start Service, on page 257.

Configuration File Corruption

Problem

If you continue to have problems with a particular phone that other suggestions in this chapter do not resolve, the configuration file may be corrupted.

Solution

Create a new phone configuration file.

Cisco Unified Communications Manager Phone Registration

Problem

The phone is not registered with the Cisco Unified Communications Manager

Solution

A Cisco IP Phone can register with a Cisco Unified Communications Manager server only if the phone is added to the server or if autoregistration is enabled. Review the information and procedures in Phone Addition Methods, on page 68 to ensure that the phone is added to the Cisco Unified Communications Manager database.

To verify that the phone is in the Cisco Unified Communications Manager database, choose **Device** > **Phone** from Cisco Unified Communications Manager Administration. Click **Find** to search for the phone based on the MAC Address. For information about determining a MAC address, see Determine the Phone MAC Address, on page 68.

If the phone is already in the Cisco Unified Communications Manager database, the configuration file may be damaged. See Configuration File Corruption, on page 245 for assistance.

Cisco IP Phone Cannot Obtain IP Address

Problem

If a phone cannot obtain an IP address when it starts up, the phone may not be on the same network or VLAN as the DHCP server, or the switch port to which the phone connects may be disabled.

Solution

Ensure that the network or VLAN to which the phone connects has access to the DHCP server, and ensure that the switch port is enabled.

Phone Not Registering

Problem

The phone screen displays the prompt "Enter activation code or service domain."

Solution

The phone is missing a TFTP address. Check that option 150 is provided by the DHCP server or an alternate TFTP is manually configured.

Phone Reset Problems

If users report that their phones are resetting during calls or while the phones are idle, you should investigate the cause. If the network connection and Cisco Unified Communications Manager connection are stable, a phone should not reset.

Typically, a phone resets if it has problems in connecting to the network or to Cisco Unified Communications Manager.

Phone Resets Due to Intermittent Network Outages

Problem

Your network may be experiencing intermittent outages.

Solution

Intermittent network outages affect data and voice traffic differently. Your network might be experiencing intermittent outages without detection. If so, data traffic can resend lost packets and verify that packets are received and transmitted. However, voice traffic cannot recapture lost packets. Rather than retransmitting a lost network connection, the phone resets and attempts to reconnect to the network. Contact the system administrator for information on known problems in the voice network.

Phone Resets Due to DHCP Setting Errors

Problem

The DHCP settings may be incorrect.

Solution

Verify that you have properly configured the phone to use DHCP. Verify that the DHCP server is set up properly. Verify the DHCP lease duration. We recommend that you set the lease duration to 8 days.

Phone Resets Due to Incorrect Static IP Address

Problem

The static IP address assigned to the phone may be incorrect.

Solution

If the phone is assigned a static IP address, verify that you have entered the correct settings.

Phone Resets During Heavy Network Usage

Problem

If the phone appears to reset during heavy network usage, it is likely that you do not have a voice VLAN configured.

Solution

Isolating the phones on a separate auxiliary VLAN increases the quality of the voice traffic.

Phone Resets Due to Intentional Reset

Problem

If you are not the only administrator with access to Cisco Unified Communications Manager, you should verify that no one else has intentionally reset the phones.

Solution

You can check if a Cisco IP Phone received a command from Cisco Unified Communications Manager to reset by pressing **Applications** on the phone and choosing **Admin settings** > **Status** > **Network statistics**.

- If the Restart Cause field displays `Reset-Reset`, the phone receives a Reset/Reset from Cisco Unified Communications Manager Administration.
- If the Restart Cause field displays `Reset-Restart`, the phone closed because it received a Reset/Restart from Cisco Unified Communications Manager Administration.

Phone Resets Due to DNS or Other Connectivity Issues

Problem

The phone reset continues and you suspect DNS or other connectivity issues.

Solution

If the phone continues to reset, eliminate DNS or other connectivity errors by following the procedure in Determine DNS or Connectivity Issues, on page 254.

Phone Does Not Power Up

Problem

The phone does not appear to be powered up.

Solution

In most cases, a phone restarts if it powers up by using external power but loses that connection and switches to PoE. Similarly, a phone may restart if it powers up by using PoE and then connects to an external power supply.

Phone Cannot Connect to LAN

Problem

The physical connection to the LAN may be broken.

Solution

Verify that the Ethernet connection to which the Cisco IP Phone connects is up. For example, check whether the particular port or switch to which the phone connects is down and that the switch is not rebooting. Also ensure that no cable breaks exist.

Cisco IP Phone Security Problems

The following sections provide troubleshooting information for the security features on the Cisco IP Phone. For information about the solutions for any of these issues, and for additional troubleshooting information about security, see *Cisco Unified Communications Manager Security Guide*.

CTL File Problems

The following sections describe troubleshooting problems with the CTL file.

Authentication Error, Phone Cannot Authenticate CTL File

Problem

A device authentication error occurs.

Cause

CTL file does not have a Cisco Unified Communications Manager certificate or has an incorrect certificate.

Solution

Install a correct certificate.

Phone Cannot Authenticate CTL File

Problem

Phone cannot authenticate the CTL file.

Cause

The security token that signed the updated CTL file does not exist in the CTL file on the phone.

Solution

Change the security token in the CTL file and install the new file on the phone.

CTL File Authenticates but Other Configuration Files Do Not Authenticate

Problem

Phone cannot authenticate any configuration files other than the CTL file.

Cause

A bad TFTP record exists, or the configuration file may not be signed by the corresponding certificate in the phone Trust List.

Solution

Check the TFTP record and the certificate in the Trust List.

ITL File Authenticates but Other Configuration Files Do Not Authenticate

Problem

Phone cannot authenticate any configuration files other than the ITL file.

Cause

The configuration file may not be signed by the corresponding certificate in the phone Trust List.

Solution

Re-sign the configuration file by using the correct certificate.

TFTP Authorization Fails

Problem

Phone reports TFTP authorization failure.

Cause

The TFTP address for the phone does not exist in the CTL file.

If you created a new CTL file with a new TFTP record, the existing CTL file on the phone may not contain a record for the new TFTP server.

Solution

Check the configuration of the TFTP address in the phone CTL file.

Phone Does Not Register

Problem

Phone does not register with Cisco Unified Communications Manager.

Cause

The CTL file does not contain the correct information for the Cisco Unified Communications Manager server.

Solution

Change the Cisco Unified Communications Manager server information in the CTL file.

Signed Configuration Files Are Not Requested

Problem

Phone does not request signed configuration files.

Cause

The CTL file does not contain any TFTP entries with certificates.

Solution

Configure TFTP entries with certificates in the CTL file.

Video Call Problems

No Video Between Two Cisco IP Video Phones

Problem

Video isn't streaming between between two Cisco IP video phones.

Solution

Check to make sure no Media Termination Point (MTP) is being used in the call flow.

Video Stutters or Drops Frames

Problem

When I'm on a video call, the video buffers or drops frames.

Solution

The quality of the image depends upon the bandwidth of the call. Raising the bit rate increases the quality of your video, but requires extra network resources. Always use the bit rate best suited for your type of video. A video call that is 720p and 15 frames per second requires a bit rate of 790 kbps or higher. A video call that is 720p and 30 frames per second requires a bit rate of 1360 kbps or higher.

For additional information about bandwidth, see the Video Transmit Resolution Setup section of the "Phone Features and Setup" chapter.

Solution

Confirm that the Maximum Session Bit Rate for Video Calls parameter is configured to be at least the minimum video bit rate range. On the Cisco Unified Communications Manager, navigate to **System** > **Region Information** > **Region**.

Cannot Transfer a Video Call

Problem

I can't transfer a video call from my desk phone to my mobile device.

Solution

Cisco Unified Mobility does not extend to video calls. A video call that is received at the desk phone cannot be picked up on the mobile phone.

No Video During a Conference call

Problem

A video call goes to an audio call when I add two or more people to the call.

You must use a video conference bridge for ad hoc and meet-me video conferencing.

General Telephone Call Problems

The following sections help troubleshoot general telephone call problems.

Phone Call Cannot Be Established

Problem

A user complains about not being able to make a call.

Cause

The phone does not have a DHCP IP address, is unable to register to Cisco Unified Communications Manager. Phones with an LCD display show the message `Configuring IP` or `Registering`. Phones without an LCD display play the reorder tone (instead of dial tone) in the handset when the user attempts to make a call.

Solution

1. Verify the following:
   a. The Ethernet cable is attached.
   b. The Cisco CallManager service is running on the Cisco Unified Communications Manager server.
   c. Both phones are registered to the same Cisco Unified Communications Manager.
2. Audio server debug and capture logs are enabled for both phones. If needed, enable Java debug.

Phone Does Not Recognize DTMF Digits or Digits Are Delayed

Problem

The user complains that numbers are missed or delayed when the keypad is used.

Cause

Pressing the keys too quickly can result in missed or delayed digits.

Solution

Keys should not be pressed rapidly.

Troubleshooting Procedures

These procedures can be used to identify and correct problems.

Create a Phone Problem Report from Cisco Unified Communications Manager

You can generate a problem report for the phones from Cisco Unified Communications Manager. This action results in the same information that the Problem Report Tool (PRT) softkey generates on the phone.

The problem report contains information about the phone and the headsets.

Procedure

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

Step 2 Click **Find** and select one or more Cisco IP Phones.

Step 3 Click **Generate PRT for Selected** to collect PRT logs for the headsets used on the selected Cisco IP Phones.

Create a Console Log from Your Phone

You generate a console log when your phone will not connect to the network and you cannot access the Problem Report Tool (PRT).

Before you begin

Connect a console cable to the Auxiliary port on the back of your phone.

Procedure

Step 1 On your phone, press **Applications** .

Step 2 Navigate **Admin settings** > **Aux port**.

Step 3 Select **Collect console log** to collect device logs.

Check TFTP Settings

Procedure

Step 1 On the Cisco IP Phone, press **Applications**, choose **Admin settings** > **Network setup** > **Ethernet setup** > **IPv4 setup** > **TFTP Server 1**.

Step 2 If you have assigned a static IP address to the phone, you must manually enter a setting for the TFTP Server 1 option.

Step 3 If you are using DHCP, the phone obtains the address for the TFTP server from the DHCP server. Check that the IP address is configured in Option 150.

Step 4 You can also enable the phone to use an alternate TFTP server. Such a setting is particularly useful if the phone recently moved from one location to another.

Step 5 If the local DHCP does not offer the correct TFTP address, enable the phone to use an alternate TFTP server.

This is often necessary in VPN scenario.

Determine DNS or Connectivity Issues

Procedure

Step 1 Use the Reset Settings menu to reset phone settings to their default values.

Step 2 Modify DHCP and IP settings:

a) Disable DHCP.

b) Assign static IP values to the phone. Use the same default router setting that other functioning phones use.

c) Assign a TFTP server. Use the same TFTP server that other functioning phones use.

Step 3 On the Cisco Unified Communications Manager server, verify that the local host files have the correct Cisco Unified Communications Manager server name mapped to the correct IP address.

Step 4 From Cisco Unified Communications Manager, choose **System** > **Server** and verify that reference to the server is made by the IP address and not by the DNS name.

Step 5 From Cisco Unified Communications Manager, choose **Device** > **Phone**. Click **Find** to search for this phone. Verify that you have assigned the correct MAC address to this Cisco IP Phone.

Step 6 Power cycle the phone.

Related Topics

Basic Reset, on page 259
Determine the Phone MAC Address, on page 68

Check DHCP Settings

Procedure

Step 1 On the phone, press **Applications** .

Step 2 Select **Wi-Fi** > **Network Setup** > **IPv4 Setup**, and look at the following options:

- DHCP Server: If you have assigned a static IP address to the phone, you do not need to enter a value for the DHCP Server option. However, if you are using a DHCP server, this option must have a value. If no value is found, check your IP routing and VLAN configuration. See the *Troubleshooting Switch Port and Interface Problems* document, available at this URL:

  http://www.cisco.com/en/US/customer/products/hw/switches/ps708/prod_tech_notes_list.html

- IP Address, Subnet Mask, Default Router: If you have assigned a static IP address to the phone, you must manually enter settings for these options.

Step 3 If you are using DHCP, check the IP addresses that your DHCP server distributes.

See the *Understanding and Troubleshooting DHCP in Catalyst Switch or Enterprise Networks* document, available at this URL:

http://www.cisco.com/en/US/tech/tk648/tk361/technologies_tech_note09186a00800f0804.shtml

Create a New Phone Configuration File

When you remove a phone from the Cisco Unified Communications Manager database, the configuration file is deleted from the Cisco Unified Communications Manager TFTP server. The phone directory number or numbers remain in the Cisco Unified Communications Manager database. They are called unassigned DNs and can be used for other devices. If unassigned DNs are not used by other devices, delete these DNs from the Cisco Unified Communications Manager database. You can use the Route Plan Report to view and delete unassigned reference numbers. For more information, see the documentation for your particular Cisco Unified Communications Manager release.

Changing the buttons on a phone button template, or assigning a different phone button template to a phone, may result in directory numbers that are no longer accessible from the phone. The directory numbers are still assigned to the phone in the Cisco Unified Communications Manager database, but the phone has no button on the phone with which calls can be answered. These directory numbers should be removed from the phone and deleted if necessary.

Procedure

Step 1 From Cisco Unified Communications Manager, choose **Device** > **Phone** and click **Find** to locate the phone that is experiencing problems.

Step 2 Choose **Delete** to remove the phone from the Cisco Unified Communications Manager database.

Note When you remove a phone from the Cisco Unified Communications Manager database, the configuration file is deleted from the Cisco Unified Communications Manager TFTP server. The phone directory number or numbers remain in the Cisco Unified Communications Manager database. They are called unassigned DNs and can be used for other devices. If unassigned DNs are not used by other devices, delete these DNs from the Cisco Unified Communications Manager database. You can use the Route Plan Report to view and delete unassigned reference numbers.

Step 3 Add the phone back to the Cisco Unified Communications Manager database.

Step 4 Power cycle the phone.

Related Topics

Cisco Unified Communications Manager Documentation, on page xv
Phone Addition Methods, on page 68

Identify 802.1X Authentication Problems

Procedure

Step 1 Verify that you have properly configured the required components.

Step 2 Confirm that the shared secret is configured on the phone.

- If the shared secret is configured, verify that you have the same shared secret on the authentication server.
- If the shared secret is not configured on the phone, enter it, and ensure that it matches the shared secret on the authentication server.

Verify DNS Settings

To verify DNS settings, follow these steps:

Procedure

Step 1 Press **Applications** .

Step 2 Select **Administrator Settings** > **Network Setup** > **IPv4 Setup** > **DNS Server 1**.

Step 3 You should also verify that a CNAME entry was made in the DNS server for the TFTP server and for the Cisco Unified Communications Manager system.

You must also ensure that DNS is configured to do reverse lookups.

Start Service

A service must be activated before it can be started or stopped.

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **Cisco Unified Serviceability** from the Navigation drop-down list and click **Go**.

Step 2 Choose **Tools** > **Control Center - Feature Services**.

Step 3 Choose the primary Cisco Unified Communications Manager server from the Server drop-down list.

The window displays the service names for the server that you chose, the status of the services, and a service control panel to start or stop a service.

Step 4 If a service has stopped, click the corresponding radio button and then click **Start**.

The Service Status symbol changes from a square to an arrow.

Control Debug Information from Cisco Unified Communications Manager

If you are experiencing phone problems that you cannot resolve, Cisco TAC can assist you. You will need to turn debugging on for the phone, reproduce the problem, turn debugging off, and send the logs to TAC for analysis.

Because debugging captures detailed information, the communication traffic can slow down the phone, making it less responsive. After you capture the logs, you should turn debugging off to ensure phone operation.

The debug information may include a single digit code that reflects the severity of the situation. Situations are graded as follows:

- 0 - Emergency
- 1 - Alert
- 2 - Critical
- 3 - Error
- 4 - Warn
- 5 - Notification
- 6 - Information
- 7 - Debugging

Contact Cisco TAC for more information and assistance.

Procedure

Step 1 In the Cisco Unified Communications Manager Administration, select one of the following windows:

- **Device** > **Device settings** > **Common Phone Profile**
- **System** > **Enterprise Phone Configuration**
- **Device** > **Phone**

Step 2 Set the following parameters:

- Log Profile - values: Preset (default), Default, Telephony, SIP, UI, Network, Media, Upgrade, Accessory, Security, Wi-Fi, VPN, Energywise, MobileRemoteAccess

  Note To implement multilevel and multi-section support of the parameters, check the Log Profile check box.

- Remote Log - values: Disable (default), Enable
- IPv6 Log Server or Log Server - IP address (IPv4 or IPv6 address)

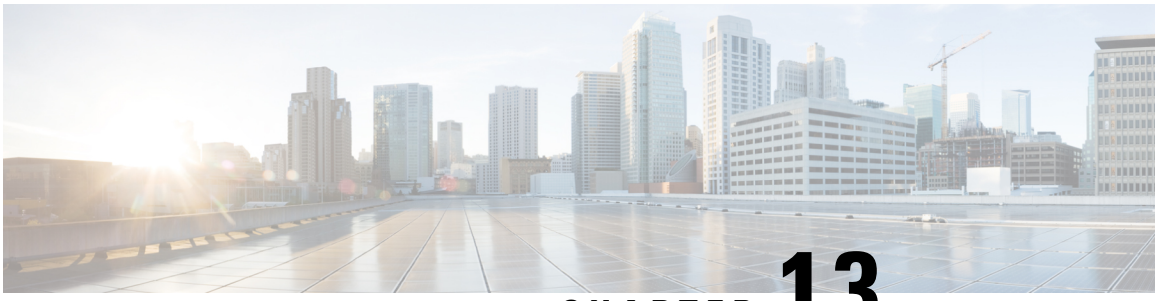
Note When the Log Server cannot be reached, the phone stops sending debug messages.

- The format for the IPv4 Log Server address is **`address:<port>@@base=<0-7>;pfs=<0-1>`**
- The format for the IPv6 Log Server address is **`[address]:<port>@@base=<0-7>;pfs=<0-1>`**
- Where:
  - the IPv4 address is separated with dot (.)
  - the IPv6 address is separated with colon (:)

Additional Troubleshooting Information

If you have additional questions about troubleshooting your phone, go to the following Cisco website and navigate to the desired phone model:

https://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8800-series/series.html#Troubleshooting



CHAPTER 13

Maintenance

- Basic Reset, on page 259
- Perform Network Configuration Reset, on page 261
- Perform User Network Configuration Reset, on page 261
- Remove CTL File, on page 261
- Quality Report Tool, on page 262
- Voice Quality Monitoring, on page 262
- Cisco IP Phone Cleaning, on page 263

Basic Reset

Performing a basic reset of a CiscoIP Phone provides a way to recover if the phone experiences an error and provides a way to reset or restore various configuration and security settings.

The following table describes the ways to perform a basic reset. You can reset a phone with any of these operations after the phone has started up. Choose the operation that is appropriate for your situation.

Table 54: Basic Reset Methods

Operation	Action	Expla
Restart phone	Press **Applications** . Go to **Admin settings** > **Reset settings** > **Reset device**.	Reset writte
Reset settings	To reset settings, press **Applications** and choose **Administrator Settings** > **Reset Settings** > **Network**.	Reset
	To reset the CTL file, press **Applications** and choose **Administrator Settings** > **Reset Settings** > **Security**.	Reset

Reset the Phone to the Factory Settings from the Phone Keypad

You can reset the phone to the factory settings. The reset clears all the phone parameters.

Procedure

Step 1 Remove power from the phone in one of these ways:

- Unplug the power adapter.
- Unplug the LAN cable.

Step 2 Wait for 5 seconds.

Step 3 Press and hold **#** and plug the phone back in. Release the **#** only when the **Headset** and **Speaker** buttons are lit.

Note In some hardware versions, the **Mute** button also lights along with **Headset** and **Speaker** buttons when you plug the phone back in. In that case, wait for all of them to go out and release **#** only when the **Headset** and **Speaker** buttons are lit again.

Step 4 Enter the following key sequence:

```
123456789*0#
```

The light for the **Headset** button turns off after you press the **1** key. After you enter the key sequence, the **Mute** button lights.

Caution Do not power down the phone until it completes the factory reset process, and the main screen appears.

The phone resets.

Perform Reset All Settings from Phone Menu

Perform this task if you want to reset your user and network setup settings to the default values.

Procedure

Step 1 Press **Applications** .

Step 2 Choose **Administrator settings** > **Reset settings** > **All settings**.

If necessary, unlock the phone options.

Reboot Your Phone from the Backup Image

Your Cisco IP Phone has a second, backup image that allows you to recover the phone when the default image has been compromised.

To reboot your phone from the backup image, perform the following procedure.

Procedure

Step 1 Disconnect the power supply.
Step 2 Press and hold the star (*) key.
Step 3 Reconnect the power. Continue pressing the star key until the Mute LED turns off.
Step 4 Release the star key.
The phone reboots from the backup image.

Perform Network Configuration Reset

Resets network configuration settings to their default values and resets the phone. This method causes DHCP to reconfigure the IP address of the phone.

Procedure

Step 1 From the Administrator Settings menu, if required, unlock phone options.
Step 2 Choose **Reset Settings** > **Network Setup**.

Perform User Network Configuration Reset

Resets any user and network configuration changes that you have made, but that the phone has not written to flash memory, to previously saved settings.

Procedure

Step 1 From the Administrator Settings menu, if required, unlock phone options.
Step 2 Choose **Reset Settings** > **Reset Device**.

Remove CTL File

Deletes only the CTL file from the phone.

Procedure

Step 1 From the Administrator Settings menu, if required, unlock phone options.

Step 2 Choose **Reset Settings** > **Security Settings**.

Quality Report Tool

The Quality Report Tool (QRT) is a voice quality and general problem-reporting tool for the Cisco IP Phone. The QRT feature is installed as part of Cisco Unified Communications Manager installation.

You can configure user Cisco IP Phones with QRT. When you do so, users can report problems with phone calls by pressing Report Quality. This softkey or button is available only when the Cisco IP Phone is in the Connected, Connected Conference, Connected Transfer, or OnHook states.

When a user presses Report Quality, a list of problem categories appears. The user selects the appropriate problem category, and this feedback is logged in an XML file. Actual information that is logged depends on the user selection and whether the destination device is a Cisco IP Phone.

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

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

Voice Quality Monitoring

To measure the voice quality of calls that are sent and received within the network, Cisco IP Phones use these statistical metrics that are based on concealment events. The DSP plays concealment frames to mask frame loss in the voice packet stream.

- Concealment Ratio metrics—Show the ratio of concealment frames over total speech frames. An interval conceal ratio is calculated every 3 seconds.
- Concealed Second metrics—Show the number of seconds in which the DSP plays concealment frames due to lost frames. A severely "concealed second" is a second in which the DSP plays more than five percent concealment frames.

Note Concealment ratio and concealment seconds are primary measurements based on frame loss. A Conceal Ratio of zero indicates that the IP network is delivering frames and packets on time with no loss.

You can access voice quality metrics from the Cisco IP Phone using the Call Statistics screen or remotely by using Streaming Statistics.

Voice Quality Troubleshooting Tips

When you observe significant and persistent changes to metrics, use the following table for general troubleshooting information.

Table 55: Changes to Voice Quality Metrics

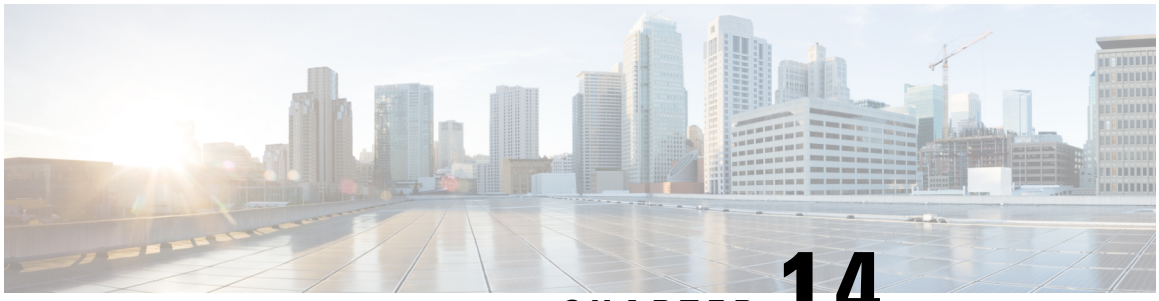
Metric Change	Condition
Conceal Ratio and Conceal Seconds increase significantly	Network impairment from packet loss or high jitter.
Conceal Ratio is near or at zero, but the voice quality is poor.	• Noise or distortion in the audio channel such as echo or audio levels. • Tandem calls that undergo multiple encode/decode such as calls to a cellular network or calling card network. • Acoustic problems coming from a speakerphone, handsfree cellular phone or wireless headset. Check packet transmit (TxCnt) and packet receive (RxCnt) counters to verify that voice packets are flowing.
MOS LQK scores decrease significantly	Network impairment from packet loss or high jitter levels: • Average MOS LQK decreases may indicate widespread and uniform impairment. • Individual MOS LQK decreases may indicate bursty impairment. Cross-check the conceal ratio and conceal seconds for evidence of packet loss and jitter.
MOS LQK scores increase significantly	• Check to see if the phone is using a different codec than expected (RxType and TxType). • Check to see if the MOS LQK version changed after a firmware upgrade.

Note Voice quality metrics do not account for noise or distortion, only frame loss.

Cisco IP Phone Cleaning

To clean your Cisco IP Phone, use only a dry soft cloth to gently wipe the phone and the phone screen. Do not apply liquids or powders directly to the phone. As with all non-weatherproof electronics, liquids and powders can damage the components and cause failures.

When the phone is in sleep mode, the screen is blank and the Select button is not lit. When the phone is in this condition, you can clean the screen, as long as you know that the phone will remain asleep until after you finish cleaning.



CHAPTER 14

International User Support

- Unified Communications Manager Endpoints Locale Installer, on page 265
- International Call Logging Support, on page 265
- Language Limitation, on page 266

Unified Communications Manager Endpoints Locale Installer

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

To access the Locale Installer required for a release, access the Software Download page, navigate to your phone model, and select the Unified Communications Manager Endpoints Locale Installer link.

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

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

Related Topics

Cisco Unified Communications Manager Documentation, on page xv

International Call Logging Support

If your phone system is configured for international call logging (calling party normalization), the call logs, redial, or call directory entries may display a plus (+) symbol to represent the international escape code for your location. Depending on the configuration for your phone system, the + may be replaced with the correct international dialing code, or you may need to edit the number before dialing to manually replace the + with the international escape code for your location. In addition, while the call log or directory entry may display the full international number for the received call, the phone display may show the shortened local version of the number, without international or country codes.

Language Limitation

There is no localized Keyboard Alphanumeric Text Entry (KATE) support for the following Asian locales:

- Chinese (Hong Kong)
- Chinese (Taiwan)
- Japanese (Japan)
- Korean (Korea Republic)

The default English (United States) KATE is presented to the user instead.

For example, the phone screen will show text in Korean, but the **2** key on the keypad will display **a b c 2 A B C**.

Chinese input works similar to PCs and mobile phones in Chinese. The Chinese locale installer is required for Chinese input to function.