



Cisco Unified IP Phone 6901 and 6911 Administration Guide for Cisco Unified Communications Manager 10.0 (SCCP and SIP)

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CONTENTS

PREFACE

| | |
|--|-----------|
| Preface | xi |
| Overview | xi |
| Audience | xi |
| Organization | xi |
| Related Documentation | xii |
| Cisco Unified IP Phone 6900 Series Documentation | xiii |
| Cisco Unified Communications Manager Documentation | xiii |
| Cisco Business Edition 3000 Documentation | xiii |
| Cisco Business Edition 5000 Documentation | xiii |
| Cisco Business Edition 6000 Documentation | xiii |
| Documentation, Support, and Security Guidelines | xiii |
| Cisco Product Security Overview | xiv |
| Guide Conventions | xiv |

CHAPTER 1

| | |
|-------------------------------|----------|
| Cisco Unified IP Phone | 1 |
| Phone Overview | 1 |
| Cisco Unified IP Phone 6901 | 2 |
| Phone Connections | 2 |
| Buttons and Hardware | 3 |
| Hookswitch Button | 4 |
| Cisco Unified IP Phone 6911 | 4 |
| Phone Connections | 4 |
| Buttons and Hardware | 6 |
| Paper Label | 7 |
| Clean the Phone Screen | 8 |
| General Phone Information | 8 |

| | |
|---|--|
| Footstand | 8 |
| Phone Display Angle | 9 |
| Raise Phone Angle | 9 |
| Lower Phone Angle | 9 |
| Important Headset Safety Information | 10 |
| Network Protocols | 10 |
| Cisco Unified IP Phone 6901 and 6911 Supported Features | 14 |
| Feature Overview | 14 |
| Telephony Feature Administration | 15 |
| Cisco Unified IP Phone Network Parameters | 15 |
| Information for End Users | 15 |
| Cisco Unified IP Phones Security Features | 16 |
| Supported Security Features | 17 |
| Security Profiles | 20 |
| Authenticated, Encrypted, and Protected Phone Calls | 20 |
| Identify Protected Calls | 21 |
| Call Security Interactions and Restrictions | 21 |
| 802.1X Authentication | 22 |
| Overview | 22 |
| Required Network Components | 23 |
| Best Practices-Requirements and Recommendations | 23 |
| Cisco Unified IP Phone Deployment | 24 |
| Cisco Unified IP Phones Setup in Cisco Unified Communications Manager | 24 |
| Set Up Cisco Unified IP Phone 6901 and 6911 in Cisco Unified Communications Manager | 24 |
| Cisco Unified IP Phones Installation | 26 |
| Install Cisco Unified IP Phone 6901 and 6911 | 26 |
| Phone Power Reduction | 28 |
| Terminology Differences | 28 |
| <hr/> | |
| CHAPTER 2 | Cisco Unified IP Phone and Telephony Networks |
| | 29 |
| Phone and Telephony Networks Overview | 29 |
| Cisco Unified IP Communications Product Interactions | 29 |
| Cisco Unified IP Phone and Cisco Unified Communications Manager Interactions | 29 |
| Cisco Unified IP Phone 6911 and VLAN Interaction | 30 |

| | |
|---|----|
| Cisco Unified IP Phone and Cisco Unified Communications Manager Express Interaction | 31 |
| Cisco Unified IP Phone Power | 32 |
| Power Guidelines | 32 |
| Power Outage | 33 |
| Additional Information About Power | 33 |
| Phone Configuration Files | 33 |
| Phone Startup Process | 34 |
| Cisco Unified Communications Manager Phone Addition Methods | 35 |
| Autoregistration Phone Addition | 36 |
| Autoregistration and TAPS Phone Addition | 37 |
| Cisco Unified Communications Manager Phone Addition | 37 |
| Add Phones Using BAT Phone Template | 38 |
| Cisco Unified IP Phones and Different Protocols | 38 |
| Convert New Phone from SCCP to SIP | 38 |
| In-Use Phone Protocol to Protocol Conversion | 39 |
| Deploy Phone in SCCP and SIP Environment | 39 |
| Cisco Unified IP Phone MAC Address Determination | 40 |

CHAPTER 3

| | |
|---|-----------|
| Cisco Unified IP Phone Installation | 41 |
| Phone Installation Overview | 41 |
| Before You Begin | 41 |
| Network Requirements | 41 |
| Cisco Unified Communications Manager Setup | 42 |
| Cisco Unified IP Phone 6901 and 6911 Components | 42 |
| Network and Access Ports | 42 |
| Handset | 43 |
| Disable Speakerphone (Cisco Unified IP Phone 6911 Only) | 43 |
| Install Cisco Unified IP Phone | 43 |
| Cisco Unified IP Phone 6901 Installation | 44 |
| Cisco Unified IP Phone 6911 Installation | 44 |
| Phone Wall Mount | 44 |
| Phone Startup Verification | 45 |
| Network Settings | 45 |
| Cisco Unified IP Phone Security | 45 |

Locally Significant Certificate Manual Setup 45

CHAPTER 4

Cisco Unified IP Phone Settings 47

Phone Settings Overview 47

Cisco Unified IP Phone Network Settings Setup 47

Phone Settings Options 47

Access Phone Configuration Settings 48

Access IVR and Set Up Phone Settings 49

IVR Configuration Menu 49

CHAPTER 5

Features, Templates, Services, and User Setup 53

Features, Templates, Services, and Users Overview 53

Telephony Features Available for Cisco Unified IP Phone 53

Add Users to Cisco Unified Communications Manager 62

User Options Web Pages Management 63

User Access to User Options Web Pages 63

Add User to End User Group 63

Associate Phones with Users 64

Customize User Options Web Pages Options 64

EnergyWise Setup on Cisco Unified IP Phone 6901 and 6911 65

UCR 2008 Setup 68

Set Up UCR 2008 in Phone 69

Set Up UCR 2008 in Common Phone Profile 69

Set Up UCR 2008 in Enterprise Phone Configuration 70

Set Up UCR 2008 in Common Device Configuration 70

CHAPTER 6

Self Care Portal Management 71

Self Care Portal Overview 71

Set Up User Access to the Self Care Portal 71

Customize the Self Care Portal Display 72

CHAPTER 7

Remote Monitoring 73

Remote Monitoring Overview 73

Cisco Unified IP Phone Web Pages 73

| | |
|------------------------------------|----|
| Access Web Page for Phone | 74 |
| Disable and Enable Web Page Access | 75 |
| Device Information Area | 75 |
| Network Setup Area | 76 |
| Network Statistics Area | 81 |
| Ethernet Information Area | 82 |
| Network Information Area | 82 |
| Device Logs Area | 83 |
| Status Messages Area | 84 |
| Streaming Statistics Area | 89 |

CHAPTER 8**Troubleshooting and Maintenance 93**

| | |
|---|-----|
| Troubleshooting and Maintenance Overview | 93 |
| Troubleshooting | 93 |
| Startup Problems | 93 |
| Cisco Unified IP Phone Does Not Go Through Normal Startup Process | 93 |
| Cisco Unified IP Phone Does Not Register with Cisco Unified Communications Manager | 94 |
| Cisco Unified IP Phone Cannot Obtain IP Address | 96 |
| Cisco Unified IP Phone Displays Flashing Red Light | 97 |
| Cisco Unified IP Phone Resets Unexpectedly | 97 |
| Physical Connection Problems | 97 |
| Intermittent Network Outages | 97 |
| DHCP Settings Errors | 98 |
| Static IP Address Settings Errors | 98 |
| Voice VLAN Setup Errors | 98 |
| Phones Have Not Been Intentionally Reset | 98 |
| DNS or Other Connectivity Errors | 99 |
| Power Connection Problems | 99 |
| Audio Problems | 99 |
| Poor Audio Quality with Calls That Route Outside Cisco Unified Communications Manager | 99 |
| Choppy Speech | 100 |
| No Speech Path | 100 |
| General Telephone Call Problems | 100 |
| Phone Call Cannot Be Established | 100 |

Phone Does Not Recognize DTMF Digits or Digits Are Delayed 101

Troubleshooting Procedures 101

 Check TFTP Settings 101

 Create New Configuration File 101

 Determine DNS or Connectivity Issues 102

 Start Service 103

General Troubleshooting Information 103

Additional Troubleshooting Information 105

Maintenance 105

 Cisco Unified IP Phone Reset or Restore 105

 Perform Basic Reset 105

 Perform Factory Reset 106

 Voice Quality Monitoring 107

 Voice Quality Metrics 107

 Voice Quality Troubleshooting Tips 108

 Cisco Unified IP Phone Cleaning 109

APPENDIX A

Internal Support Web Site 111

 Internal Web Site Overview 111

 Cisco Unified IP Phone User Support 111

 User Options Web Pages Access 111

 Phone Features User Subscription and Setup 112

 User Voice Messaging System Access 112

APPENDIX B

International User Support 113

 International User Support Overview 113

 Unified Communications Manager Endpoints Locale Installer 113

APPENDIX C

Technical Specifications 115

 Physical and Operating Environment Specifications 115

 Physical and Operating Environment Specifications for Cisco Unified IP Phone 6901 115

 Physical and Operating Environment Specifications for Cisco Unified IP Phone 6911 116

 Cable Specifications 117

 Network and Access Port Pinouts 117

| | |
|---|-----|
| Network Port Connector | 117 |
| Computer Port Connector | 118 |
| Phone Behavior During Times of Network Congestion | 119 |

| | | |
|-------------------|---|------------|
| APPENDIX D | Basic Phone Administration Steps | 121 |
| | Phone Administration Overview | 121 |
| | Example User Information | 121 |
| | Cisco Unified Communications Manager User Addition | 122 |
| | Add User from External LDAP Directory | 122 |
| | Add User Directly to Cisco Unified Communications Manager | 122 |
| | Phone Setup | 123 |
| | Perform Final End User Configuration Steps | 126 |

| | | |
|-------------------|---|------------|
| APPENDIX E | Cisco Unified IP Phone Wall Mount | 129 |
| | Wall Mount Components for Cisco Unified IP Phone 6901 | 129 |
| | Install Phone on Wall Mount Plate | 130 |
| | Wall Mount Components for Cisco Unified IP Phone 6911 | 132 |
| | Before You Begin | 132 |
| | Install Bracket | 133 |
| | Adjust Handset Rest | 137 |

| | | |
|-------------------|---|------------|
| APPENDIX F | Cisco Unified IP Phone Non-Lockable Wall Mount | 139 |
| | ADA Non-Lockable Wall Mount Kit for 6900 Series | 139 |
| | Components | 140 |
| | Before You Begin | 141 |
| | Install Non-Lockable Wall Mount Kit for Phone | 141 |
| | Remove Phone from Non-Lockable Wall Mount | 144 |

| | | |
|-------------------|------------------------------------|------------|
| APPENDIX G | Feature Support by Protocol | 147 |
| | Feature Support | 147 |



Preface

- [Overview](#), on page xi
- [Audience](#), on page xi
- [Organization](#), on page xi
- [Related Documentation](#), on page xii
- [Documentation, Support, and Security Guidelines](#), on page xiii
- [Guide Conventions](#), on page xiv

Overview

Cisco Unified IP Phone 6901 and 6911 Administration Guide for Cisco Unified Communications Manager (SCCP and SIP) provides the information you need to understand, install, configure, manage, and troubleshoot the phones on a Voice over IP (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.

Related Topics

[Related Documentation](#), on page xii

Audience

Network engineers, system administrators, and telecom engineers should review this guide to learn the steps that are required to set up Cisco Unified 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.

Organization

This manual is organized as follows:

| Chapter | Description |
|---------|-------------|
|---------|-------------|

| | |
|---|---|
| Cisco Unified IP Phone, on page 1 | Provides a conceptual overview and description of the Cisco Unified IP Phone. |
| Cisco Unified IP Phone and Telephony Networks, on page 29 | Describes how the Cisco Unified IP Phone interacts with other key IP telephony components, and provides an overview of the tasks required prior to installation. |
| Cisco Unified IP Phone Installation, on page 41 | Describes how to properly and safely install and configure the Cisco Unified IP Phone on your network. |
| Cisco Unified IP Phone Settings, on page 47 | Describes how to configure network settings, verify status, and make global changes to the Cisco Unified IP Phone. |
| Features, Templates, Services, and User Setup, on page 53 | Provides an overview of procedures for configuring telephony features, configuring directories, configuring phone button, and adding users to Cisco Unified Communications Manager. |
| Remote Monitoring, on page 73 | Describes the information that you can obtain from the phone's web page to remotely monitor the operation of a phone and to assist with troubleshooting. |
| Troubleshooting and Maintenance, on page 93 | Provides tips for troubleshooting the Cisco Unified IP Phone. |
| Internal Support Web Site, on page 111 | Provides suggestions for setting up a website for providing users with important information about their Cisco Unified IP Phones. |
| International User Support, on page 113 | Provides information about setting up phones in non-English environments. |
| Technical Specifications, on page 115 | Provides technical specifications of the Cisco Unified IP Phone. |
| Basic Phone Administration Steps, on page 121 | Provides procedures for basic administration tasks such as adding a user and phone to Cisco Unified Communications Manager and then associating the user to the phone. |
| Cisco Unified IP Phone Wall Mount, on page 129 | Contains instructions for installing the wall mount for the Cisco Unified IP Phone. |
| Cisco Unified IP Phone Non-Lockable Wall Mount, on page 139 | Contains instructions for installing the Cisco Unified IP Phone Non-Lockable Wall Mount for the Cisco Unified IP Phone 6911. |
| Feature Support by Protocol, on page 147 | Provides information about feature support for the Cisco Unified IP Phone. |

Related Documentation

Use the following sections to obtain related information.

Cisco Unified IP Phone 6900 Series Documentation

Refer to publications that are specific to your language, phone model and Cisco Unified Communications Manager release. Navigate from the following documentation URL:

<https://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-6900-series/tsd-products-support-series-home.html>

Cisco Unified Communications Manager Documentation

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

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

Cisco Business Edition 3000 Documentation

See the *Cisco Business Edition 3000 Documentation Guide* and other publications that are specific to your Cisco Business Edition 3000 release. Navigate from the following documentation URL:

<https://www.cisco.com/c/en/us/support/unified-communications/business-edition-3000/tsd-products-support-series-home.html>

Cisco Business Edition 5000 Documentation

See the *Cisco Business Edition 5000 Documentation Guide* and other publications that are specific to your Cisco Business Edition 5000 release. Navigate from the following URL:

<https://www.cisco.com/c/en/us/support/unified-communications/business-edition-5000/tsd-products-support-series-home.html>

Cisco Business Edition 6000 Documentation

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

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

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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Cisco Product Security Overview

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

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

Guide Conventions

This document uses the following conventions:

| Convention | Description |
|---------------------------|--|
| boldface font | Commands and keywords are in boldface . |
| <i>italic</i> font | Arguments for which you supply values are in <i>italics</i> . |
| [] | Elements in square brackets are optional. |
| { 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 will include the quotation marks. |
| screen font | Terminal sessions and information the system displays are in <code>screen</code> font. |
| input font | Information you must enter is in input font. |
| <i>italic screen</i> font | Arguments for which you supply values are in <i>italic screen</i> font. |
| ^ | The symbol ^ represents the key labeled Control - for example, the key combination ^D in a screen 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



CHAPTER 1

Cisco Unified IP Phone

- [Phone Overview, on page 1](#)
- [Cisco Unified IP Phone 6901, on page 2](#)
- [Cisco Unified IP Phone 6911, on page 4](#)
- [General Phone Information, on page 8](#)
- [Network Protocols, on page 10](#)
- [Cisco Unified IP Phone 6901 and 6911 Supported Features, on page 14](#)
- [Cisco Unified IP Phones Security Features, on page 16](#)
- [Cisco Unified IP Phone Deployment, on page 24](#)
- [Phone Power Reduction, on page 28](#)
- [Terminology Differences, on page 28](#)

Phone Overview

The Cisco Unified IP Phone 6901 and 6911 provides voice communication over an Internet Protocol (IP) network. The Cisco Unified IP Phone functions much like a digital business phone, allowing you to place and receive phone calls. The Cisco Unified IP Phones in this guide have the following attributes:

- Cisco Unified IP Phone 6901 supports basic features such as hold, redial, transfer, and conference.
- Cisco Unified IP Phone 6911 supports advanced features such as mute, hold, transfer, conference, speed dial, call forward, and more.

A Cisco Unified IP Phone, like other network devices, must be configured and managed. These phones encode G.711a, G.711mu, G.729a, G.729ab, and iLBC codecs, and decode G.711a, G.711mu, G.729, G.729ab, and iLBC codecs. These phones encode and decode the codecs in similar manner.



Caution

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

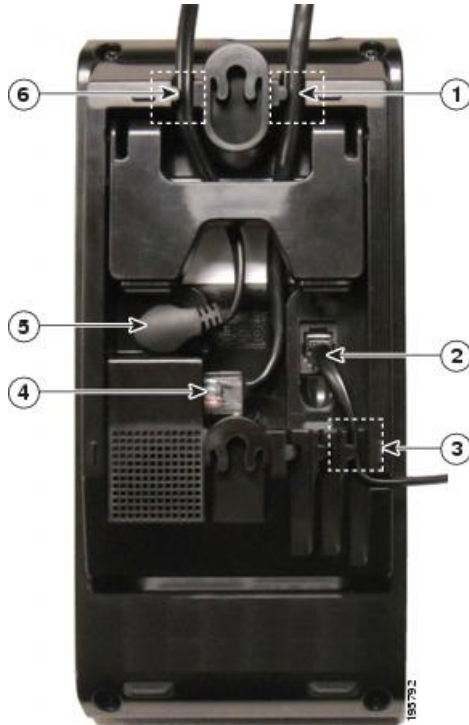
For more information, see the *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.

Cisco Unified IP Phone 6901

The following sections describe the Cisco Unified IP Phone 6901 hardware.

Phone Connections



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





| | | | |
|---|--------------------------|---|--|
| 1 | Slot for Ethernet cable. | 4 | Network port (10/100 SW) connection. IEEE 802.3af power enabled. |
| 2 | Handset connection. | 5 | DC adaptor port (DC48V). |
| 3 | Slot for handset cable. | 6 | Slot for DC adaptor cable. |

Buttons and Hardware



| | | |
|---|--|---|
| 1 | Hookswitch | Activates the features (hookflash) on your phone. |
| 2 | Hold button  | Places a connected call on hold. |
| 3 | Redial button  | Dials the last dialed number. |

| | | |
|---|--|--|
| 4 | <p>Line button</p>  | <p>Allows you to pick up a second incoming call. The Line button LED indicates the call status.</p> <p>Allows you to answer a ringing call and swap between two calls on the same line. Also, you can use the line button to create a new call when the phone is idle. The LED associated with the line button lights up to reflect the line status.</p> <p>Color LEDs indicate the line state:</p> <ul style="list-style-type: none"> • Green, steady—Active call • Green, flashing—Held call • Amber, Flashing—Incoming call • Amber, steady—Call Forward All activated • Red, steady—Remote line in use (shared line) • Red, flashing—Remote line on hold |
| 5 | <p>Volume button</p>  | <p>Controls the handset (off hook) and the ringer volume (on hook).</p> |
| 6 | <p>Keypad</p> | <p>Allows you to dial phone numbers.</p> |
| 7 | <p>Handset with light strip</p> | <p>Lights up to indicate a ringing call (flashing red) or a new voice message (steady red).</p> |

Hookswitch Button

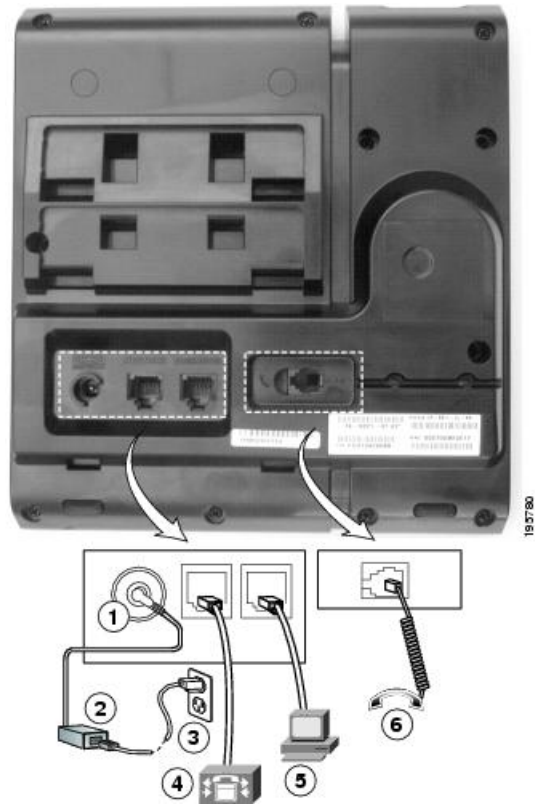
The cradle rest of your phone contains the hookswitch button. You can press and quickly release the hookswitch button to activate features (hookflash) on your phone.

Cisco Unified IP Phone 6911

The following sections describe the Cisco Unified IP Phone 6901 hardware.

Phone Connections






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








| | | | |
|---|--------------------------------|---|--|
| 1 | | 4 | Network port (10/100 SW) connection. IEEE 802.3af power enabled. |
| 2 | | 5 | |
| 3 | AC power wall plug (optional). | 6 | Handset connection. |

Buttons and Hardware



| | | |
|---|--|---|
| 1 | Handset with light strip | Lights up to indicate a ringing call (flashing red) or a new voice message (steady red). |
| 2 | Paper label | A paper strip used to enter name and contact numbers. |
| 3 | Transfer button  | Transfers a call. |
| 4 | Conference button  | Creates a conference call. |
| 5 | Hold button  | Places an active call on hold. |
| 6 | Line button  | Allows you to pick up a second incoming call and to resume a held call. The LED shows call status. |
| 7 | Speakerphone button  | Selects the speakerphone as the default audio path and initiates a new call, picks up an incoming call, or ends a call. During a call, the button lights green. The speakerphone audio path does not change until you select a new audio path (for example, by picking up the handset). |

| | | |
|----|--|--|
| 8 | Keypad | Allows you to dial phone numbers. |
| 9 | Mute button  | Toggles the microphone on or off. When the microphone is muted, the button lights red. |
| 10 | Volume button  | Controls the handset and speakerphone volume (off hook) and the ringer volume (on hook). |
| 11 | Messages button  | Auto dials your voice messaging system. |
| 12 | Redial button  | Dials the last dialed number. |
| 13 | Feature button  | <p>Depending on the phone setup, the feature button provides you with access to Speed Dial, Call Forward All, Pickup, Group Pickup and Meet Me features. You can configure up to nine items on the feature button. To access these features, press the feature button followed by the number associated with the feature. You must press the feature button and the number within five seconds of each other. The number can only be a single digit number from 1–9.</p> <p>You can access the following features either off hook or on hook:</p> <ul style="list-style-type: none"> • Call Forward All—Allows you to forward a call. • Pickup—Allows you to pickup a call on the third party phone. • Group Pickup—Allows you pick up a call within a group. • Meet Me—Allows you setup a conference. |
| 14 | Handset | Phone handset. |

Paper Label

Cisco Unified IP Phone 6911 does not include an LCD display. Cisco provides a paper strip, which can be used to enter name and contact numbers.

Clean the Phone Screen

Procedure

If your phone screen gets dirty, wipe it with a soft, dry cloth.

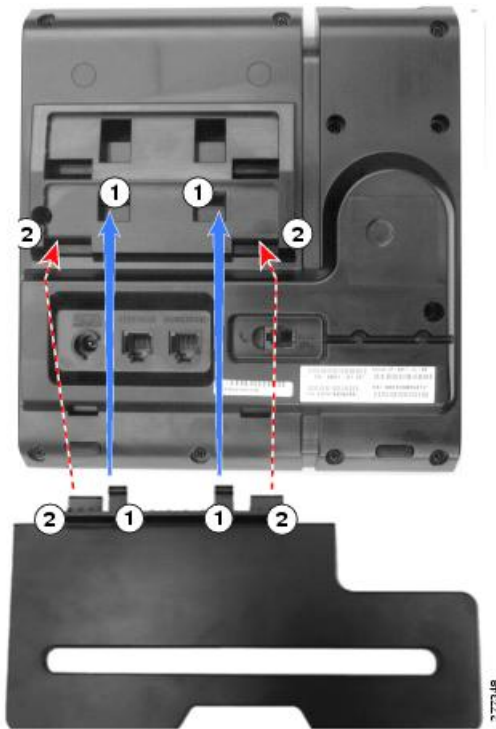
Caution Do not use any liquids or powders on the phone because they can contaminate the phone components and cause failures.

General Phone Information

This section contains information that is common to all the IP Phone models in this guide.

Footstand

If the phone is placed on a table or desk, the footstand can be connected to the back of your phone for a higher or lower viewing angle, depending on your preference.



| | | | |
|---|---|---|--|
| 1 | Insert the connectors into the lower slots. | 2 | Lift the footstand until the connectors snap into the upper slots. |
|---|---|---|--|

Phone Display Angle

Raise Phone Angle

Procedure

Connect the footstand to the lower slots for a higher viewing angle, as shown in the following figure.



Lower Phone Angle

Procedure

Connect the footstand to the upper slots for a lower viewing angle, as shown in the following figure.



Important Headset Safety Information



High Sound Pressure—Avoid listening to high volume levels for long periods to prevent possible hearing damage.

When you plug in your headset, lower the volume of the headset speaker before you put the headset on. If you remember to lower the volume before you take the headset off, the volume will start lower when you plug in your headset again.

Be aware of your surroundings. When you use your headset, it may block out important external sounds, particularly in emergencies or in noisy environments. Don't use the headset while driving. Don't leave your headset or headset cables in an area where people or pets can trip over them. Always supervise children who are near your headset or headset cables.

Network Protocols

Cisco Unified IP Phones support several industry-standard and Cisco network protocols required for voice communication. The following table provides an overview of the network protocols that the Cisco Unified IP Phone 6901 and 6911 support.

Table 1: Supported Network Protocols on the Cisco Unified IP Phone

| Network Protocol | Purpose | Usage notes |
|--|---|--|
| Cisco Audio Session Tunneling (CAST) (Cisco Unified IP Phone 6911 only) | The CAST protocol allows IP Phones and associated applications behind the phone to discover and communicate with the remote endpoints without requiring changes to the traditional signaling components like Cisco Unified Communications Manager and gateways. The CAST protocol allows separate hardware devices to synchronize related media and it allows PC applications to augment nonvideo-capable phones to become video enabled by using the PC as the video resource. | - |
| Cisco Discovery Protocol (CDP) | CDP is a device-discovery protocol that runs on all Cisco-manufactured equipment. Using CDP, a device advertises its existence to other devices and receives information about other devices in the network. | The Cisco Unified IP Phone uses 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. |
| 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 have the phone become operational without needing to manually assign an IP address or to configure additional network parameters. | By default, the phone has DHCP enabled. If disabled, you must manually configure the IP address, subnet mask, gateway, and a TFTP server on each phone locally. Cisco recommends that you use DHCP custom option 150. With this method, you configure the TFTP server IP address as the option value. For additional supported DHCP configurations, go to the “Dynamic Host Configuration Protocol” chapter and the “Cisco TFTP” chapter in the <i>Cisco Unified Communications Manager System Guide</i> . 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 Unified IP Phones use HTTP for troubleshooting purposes. |
| IEEE 802.1X | <p>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.</p> <p>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 connects. After successful authentication, normal traffic passes through the port.</p> | <p>The Cisco Unified IP Phone implements the IEEE 802.1X standard by providing support for the following authentication methods: EAP-FAST and EAP-TLS.</p> <p>When 802.1X authentication is enabled on the phone, you should disable the voice VLAN. Refer to the 802.1X Authentication, on page 22 for additional information.</p> |
| Internet Protocol (IP) | IP is a messaging protocol that addresses and sends packets across the network. | <p>To communicate using IP, network devices must have an assigned IP address, subnet, and gateway.</p> <p>IP addresses, subnets, and gateways are automatically assigned if you are using the Cisco Unified IP Phone with Dynamic Host Configuration Protocol (DHCP). If you are not using DHCP, you must manually assign these properties to each phone locally.</p> <p>The Cisco Unified IP Phones support IPv6 address. For more information, see “Internet Protocol Version 6 (IPv6)” in the <i>Cisco Unified Communications Manager Features and Services Guide</i>.</p> |
| Link Layer Discovery Protocol (LLDP) (Cisco Unified IP Phone 6911 only) | LLDP is a standardized network discovery protocol (similar to CDP) that is supported on some Cisco and third-party devices. | The Cisco Unified IP Phone 6911 supports LLDP on the switch and 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 developed for voice products. | <p>The Cisco Unified IP Phone supports LLDP-MED on the SW port to communicate information such as:</p> <ul style="list-style-type: none"> • Voice VLAN configuration • Device discovery • Power management • Inventory management <p>For more information about LLDP-MED support, see the LLDP-MED and Cisco Discovery Protocol white paper:</p> <p>http://www.cisco.com/.../whitepapers/lldp-med-and-cdp-white-paper.pdf</p> |
| Real-Time Transport Protocol (RTP) | RTP is a standard protocol for transporting real-time data, such as interactive voice and video, over data networks. | Cisco Unified 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. | By default, the phones have RTCP disabled, but you can enable it on each individual phone using Cisco Unified Communications Manager. |
| 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. | <p>Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.</p> <p>You can configure the Cisco Unified IP Phone to use either SIP or Skinny Client Control Protocol (SCCP).</p> |
| Skinnny Client Control Protocol (SCCP) | SCCP includes a messaging set that allows communications between call control servers and endpoint clients such as IP Phones. SCCP is proprietary to Cisco Systems. | Cisco Unified IP Phone 6901 and 6911 use SCCP, version 20 for call control. |

| Network Protocol | Purpose | Usage notes |
|---------------------------------------|---|---|
| Transmission Control Protocol (TCP) | TCP is a connection-oriented transport protocol. | Cisco Unified IP Phones use TCP to connect to Cisco Unified Communications Manager. |
| Transport Layer Security (TLS) | TLS is a standard protocol for securing and authenticating communications. | When security is implemented, Cisco Unified IP Phones use the TLS protocol when securely registering with Cisco Unified Communications Manager. For more information, see the <i>Cisco Unified Communications Manager Security Guide</i> . |
| Trivial File Transfer Protocol (TFTP) | TFTP allows you to transfer files over the network. On the Cisco Unified IP Phone, TFTP enables you to obtain a configuration file specific to the phone type. | TFTP requires a TFTP server in your network, which can be automatically identified from the DHCP server. If you want a phone to use a TFTP server other than the one specified by the DHCP server, you must manually assign the IP address of the TFTP server by using the Network Configuration menu on the phone. For more information, go to the “Cisco TFTP” chapter in the <i>Cisco Unified Communications Manager System Guide</i> . |
| User Datagram Protocol (UDP) | UDP is a connectionless messaging protocol for delivery of data packets. | Cisco Unified IP Phones transmit and receive RTP streams, which use UDP. |

Related Topics

[Cisco Unified IP Communications Product Interactions](#), on page 29
[Phone Startup Process](#), on page 34

Cisco Unified IP Phone 6901 and 6911 Supported Features

Cisco Unified IP Phones function much like a digital business phone, allowing you to place and receive phone calls. In addition to traditional telephony features, the Cisco Unified IP Phone includes features that enable you to administer and monitor the phone as a network device.

Feature Overview

Cisco Unified IP Phones provide traditional telephony functionality, Call Forward, Transfer, Redial, Conference, and voice message system access. Cisco Unified IP Phones also provide a variety of other features. For more

information on the features, see the *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.

As with other network devices, you must configure Cisco Unified 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, but if your network requires it, you can manually configure information, such as an IP address, TFTP server, and subnet information.

Finally, because the Cisco Unified IP Phone is a network device, you can obtain detailed status information from the phone directly. This information can assist you with troubleshooting any problems users might encounter when using their IP Phones.

Related Topics

[Cisco Unified IP Phone Settings](#), on page 47

[Features, Templates, Services, and User Setup](#), on page 53

[Troubleshooting and Maintenance](#), on page 93

Telephony Feature Administration

You can modify additional settings for the Cisco Unified IP Phone from Cisco Unified Communications Manager. Use Cisco Unified Communications Manager to set up phone registration criteria and calling search spaces, among other tasks.

For more information about Cisco Unified Communications Manager, see the Cisco Unified Communications Manager documentation, including *Cisco Unified Communications Manager Administration Guide*. You can also use the context-sensitive help available within the application for guidance.

Related Topics

[Telephony Features Available for Cisco Unified IP Phone](#), on page 53

[Related Documentation](#), on page xii

Cisco Unified IP Phone Network Parameters

You configure parameters such as DHCP, TFTP, and IP settings on the phone.

Related Topics

[Cisco Unified IP Phone Settings](#), on page 47

Information for End Users

If you are a system administrator, you are likely the primary source of information for Cisco Unified IP Phone users in your network or company. To ensure that you distribute the most current feature and procedural information, familiarize yourself with Cisco Unified IP Phone documentation on the Cisco Unified IP Phone web site.

In addition to providing documentation, you should inform users about the features enabled on the phones, including those specific to your company or network. You should also give users instructions on how to access and customize the features, if appropriate. You might want to use an internal support web site to keep your users informed.

Related Topics

[Internal Support Web Site](#), on page 111

[Cisco Unified IP Phone 6900 Series Documentation](#), on page xiii

Cisco Unified IP Phones Security Features

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 authenticated and 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 Unified IP Phones.

The Cisco Unified IP Phone 6901 and 6911 use the Phone security profile, which defines whether the device is nonsecure, authenticated, or encrypted. For information on applying the security profile to the phone, see the *Cisco Unified Communications Manager Security Guide*.

If you configure security-related settings in Cisco Unified Communications Manager, 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 “Configuring Encrypted Phone Configuration Files” chapter in *Cisco Unified Communications Manager Security Guide*.

The following table shows where you can find additional information about security in this and other documents.

Table 2: Cisco Unified IP Phone and Cisco Unified Communications Manager Security Topics

| Topic | Reference |
|---|---|
| Detailed explanation of security, including set up, configuration, and troubleshooting information for Cisco Unified Communications Manager and Cisco Unified IP Phones | <i>Troubleshooting Guide for Cisco Unified Communications Manager</i> and <i>Cisco Unified Communications Manager Security Guide</i> |
| Security features supported on the Cisco Unified IP Phone | Supported Security Features, on page 17 |
| Viewing a security profile name | See Supported Security Features, on page 17 for an overview of the security features supported by the Cisco Unified IP Phone 6901 and 6911. For more information about these features and about Cisco Unified Communications Manager and Cisco Unified IP Phone security, refer to the <i>Cisco Unified Communications Manager Security Guide</i> . |
| Identifying phone calls for which security is implemented | Authenticated, Encrypted, and Protected Phone Calls, on page 20 |
| TLS connection | <ul style="list-style-type: none"> • Network Protocols, on page 10 • Cisco Unified Communications Manager Phone Addition Methods, on page 35 |
| Security and the phone startup process | Phone Startup Process, on page 34 |

| Topic | Reference |
|---|---|
| Security and phone configuration files | Cisco Unified Communications Manager Phone Addition Methods, on page 35 |
| Disabling access to phone web pages | Disable and Enable Web Page Access, on page 75 |
| Troubleshooting | <ul style="list-style-type: none"> • Troubleshooting and Maintenance, on page 93 • Troubleshooting Guide for Cisco Unified Communications Manager |
| Deleting the CTL file from the phone | Cisco Unified IP Phone Reset or Restore, on page 105 |
| Resetting or restoring the phone | Cisco Unified IP Phone Reset or Restore, on page 105 |
| 802.1X Authentication for Cisco Unified IP Phones | <ul style="list-style-type: none"> • 802.1X Authentication, on page 22 • Troubleshooting and Maintenance, on page 93 |

Supported Security Features

The following table provides an overview of the security features that the Cisco Unified IP Phone 6901 and 6911 support. For more information about these features and about Cisco Unified Communications Manager and Cisco Unified IP Phone security, see the *Cisco Unified Communications Manager Security Guide*.



Note Most security features are available only if the phone contains a certificate trust list (CTL). For more information about the CTL, see the “Configuring the Cisco CTL Client” chapter in *Cisco Unified Communications Manager Security Guide*.

Table 3: Overview of Security Features

| Feature | Description |
|--|--|
| Image authentication | Signed binary files (with the extension .zz.sgn) prevent tampering with the firmware image before it 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 Unified IP Phone requires a unique certificate for device authentication. Phones include a manufacturing installed certificate (MIC), but for additional security, you can specify in Cisco Unified Communications Manager Administration that a certificate be installed by using the Certificate Authority Proxy Function (CAPF). |

| Feature | Description |
|-------------------------------------|---|
| 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 will not register phones unless they can be authenticated by the Cisco Unified Communications Manager. |
| 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 the file creation. Files that fail authentication are not written to flash memory on the phone. The phone rejects unauthenticated files without further processing. |
| Signaling Authentication | Uses the TLS protocol to validate that no tampering has occurred to signaling packets during transmission. |
| Manufacturing installed certificate | Each Cisco Unified IP Phone contains a unique manufacturing installed certificate (MIC), which is used for device authentication. The MIC is a permanent unique proof of identity for the phone, and allows Cisco Unified Communications Manager to authenticate the phone. |
| Secure SRST reference | <p>After you configure a Cisco Unified Survivable Remote Site Telephony (SRST) reference for security and reset the dependent devices in Cisco Unified Communications Manager, the TFTP server adds the SRST certificate to the phone configuration file and sends the file to the phone. A secure phone uses a TLS connection to interact with the SRST-enabled router.</p> <p>The configuration file uses one of the following extensions:</p> <ul style="list-style-type: none"> • .cnf.xml • .cnf.xml.sgn • .cnf.xml.enc.sgn |
| Media encryption | Uses SRTP to ensure that the media streams between supported phones proves secure and that only the intended phone receives and reads the data. Includes creating a media primary key pair for the phones, delivering the keys to the phones, and securing the delivery of the keys while the keys are in transport. |

| Feature | Description |
|--|--|
| Signaling encryption | Ensures that all SCCP and SIP signaling messages sent between the phone and the Cisco Unified Communications Manager server are encrypted. |
| Certificate Authority Proxy Function (CAPF) | 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 profiles | Defines whether the phone is nonsecure, authenticated, encrypted, or protected. |
| Encrypted configuration files | Ensures the privacy of phone configuration files. |
| Optional disabling of the web server functionality for a phone | Prevents access to a phone web page that displays a variety of operational statistics for the phone. |
| Phone hardening | The following is an additional security option that you control from Cisco Unified Communications Manager: <ul style="list-style-type: none"> Disabling access to web pages for a phone |
| 802.1X Authentication | Defines the use of 802.1X authentication to request and gain access to the network. |
| Voice Quality Metrics | |
| MOS LQK | Objective estimate of the Mean Opinion Score (MOS) for Listening Quality (LQK) that ranks audio quality from 5 (excellent) to 1 (bad). This score is based on audible-concealment events due to a frame loss in the preceding 8 seconds of the voice stream. <p>Note The MOS LQK score can vary based on the type of codec that the Cisco Unified IP Phone uses.</p> |
| Avg MOS LQK | Average MOS LQK score for the entire voice stream. |
| Min MOS LQK | Lowest MOS LQK score from the start of the voice stream. |

| Feature | Description |
|-----------------|--|
| Max MOS LQK | <p>Baseline or highest MOS LQK score from the start of the voice stream.</p> <p>The following codecs provide the corresponding maximum MOS LQK scores under normal conditions with no frame loss:</p> <ul style="list-style-type: none"> • G.711: 4.5 • G.728/iLBC: 3.9 • G729A/AB: 3.7 |
| MOS LQK Version | Version of the Cisco-proprietary algorithm used to calculate the MOS LQK scores. |

Related Topics

- [Cisco Unified IP Phones Security Features](#), on page 16
- [Security Profiles](#), on page 20
- [802.1X Authentication](#), on page 22
- [Authenticated, Encrypted, and Protected Phone Calls](#), on page 20
- [Cisco Unified IP Phone Security](#), on page 45

Security Profiles

All Cisco Unified IP Phones that support Cisco Unified Communications Manager use a security profile, which defines whether the phone is nonsecure, authenticated, or encrypted. For information about configuring the security profile and applying the profile to the phone, see the *Cisco Unified Communications Manager Security Guide*.

To view the phone security mode, you can view the security profile in Cisco Unified Communications Manager.

Related Topics

- [Authenticated, Encrypted, and Protected Phone Calls](#), on page 20

Authenticated, Encrypted, and Protected Phone Calls

In an authenticated call, all devices participating in the establishment of the call are trusted devices, and authenticated by Cisco Unified Communications Manager.

In an encrypted call, all devices participating in the establishment of the call are trusted devices, and authenticated by Cisco Unified Communications Manager. In addition, call signaling and media streams are encrypted. An encrypted call offers a high level of security, providing integrity and privacy to the call.

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.

In a protected call, a security tone plays at the beginning of a call to indicate that the other connected phone is also receiving and transmitting encrypted audio and video (if video is involved). If your call is connected to a non-protected phone, the security tone does not play.



Note Connections between two phones support protected calling. Some features, such as conference calling and shared lines, are not available when protected calling is configured. Protected calls are not authenticated.

Related Topics

[Cisco Unified IP Phones Security Features](#), on page 16

Identify Protected Calls

A protected call establishes when your phone and the phone on the other end are set up for protected calling. The other phone can be in the same Cisco IP network, or on a network outside the IP network. Protected calls can only be made between two phones. Conference calls and other multiple-line calls do not support protected calls.

A protected call establishes using this process:

1. A user initiates the call from a protected phone (protected security mode).
2. A security tone plays if the call connects to another protected phone, indicating that both ends of the conversation are encrypted and protected. If the call connects to a non protected phone, then the secure tone does not play.



Note Connections between two phones support protected calling. Some features, such as conference calling and shared lines are not available when protected calling is configured.

Call Security Interactions and Restrictions

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

The following table provides information about changes to call security levels when using Barge for Cisco Unified IP Phone 6911.

Table 4: Call Security Interactions When Using Barge (Cisco Unified IP Phone 6911 Only)

| Initiator's phone security level | Feature used | Call security level | Results of action |
|----------------------------------|--------------|---------------------|--|
| Nonsecure | cBarge | Encrypted call | Call barged and identified as nonsecure call |
| Secure (encrypted) | cBarge | Authenticated call | Call barged and identified as authenticated call |
| Secure (authenticated) | cBarge | Encrypted call | Call barged and identified as authenticated call |
| Nonsecure | cBarge | Authenticated call | Call barged and identified as nonsecure call |

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 5: Security Restrictions with Conference Calls

| Initiator's phone security level | Feature used | Security level of participants | Results of action |
|-------------------------------------|--------------|---|---|
| Nonsecure | Conference | Encrypted or authenticated | Nonsecure conference bridge Nonsecure conference |
| Secure (encrypted or authenticated) | Conference | At least one member is nonsecure | Nonsecure conference |
| Secure (encrypted) | Conference | All participants are encrypted | Secure encrypted level conference |
| Secure (authenticated) | Conference | All participants are encrypted or authenticated | Secure authenticated level conference |
| Nonsecure | cBarge | All participants are encrypted | Conference changes to nonsecure |
| Nonsecure | Meet Me | Minimum security level is encrypted | Initiator receives the message Does not meet Security Level, and the call rejected. |
| Secure (encrypted) | Meet Me | Minimum security level is authenticated | Conference accepts encrypted and authenticated calls |
| Secure (encrypted) | Meet Me | Minimum security level is nonsecure | Only secure conference bridge available and used Conference accepts all calls |

802.1X Authentication

The following sections describe the 802.1X support on the Cisco Unified IP Phones.

Overview

Cisco Unified 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 Unified IP Phones provide an EAPOL pass-through mechanism. This mechanism allows a workstation attached to the Cisco Unified 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 Unified 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.

Cisco Unified IP Phones also 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, EAP-TLS, and EAP-MD5 options for network authentication.

Required Network Components

Support for 802.1X authentication on Cisco Unified IP Phones requires several components, including:

- Cisco Unified IP Phone: The phone acts as the 802.1X *supplicant*, which initiates the request to access the network.
- 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.

Best Practices-Requirements and Recommendations

- Enable 802.1X Authentication: If you want to use the 802.1X standard to authenticate Cisco Unified IP Phones, be sure that you have properly configured the other components before enabling the standard on the phone.
- Configure PC Port: The 802.1X standard does not take into account the use of VLANs and thus Cisco recommends that only a single device should be authenticated to a specific switch port. However, some switches (including Cisco Catalyst switches) support multi-domain authentication. The switch configuration determines whether you can connect a PC to the PC port of the phone.



Note Only Cisco Unified IP Phone 6911 has a PC port.

- Enabled: If you are using a switch that supports multi-domain authentication, you can enable the PC port and connect a PC to it. In this case, Cisco Unified 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, refer to 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 will deny 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 multi-domain authentication, you can continue to use the Voice VLAN.
- Disabled: If the switch does not support multi-domain authentication, disable the Voice VLAN and consider assigning the port to the native VLAN.

Cisco Unified IP Phone Deployment

When deploying 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 “System Configuration Overview” chapter in *Cisco Unified Communications Manager System Guide*.

After you set up the IP telephony system and configure system-wide features in Cisco Unified Communications Manager, you can add IP Phones to the system.

Cisco Unified IP Phones Setup in Cisco Unified Communications Manager

To add phones to the Cisco Unified Communications Manager database, you can use:

- Autoregistration: Not supported if Cisco Unified Communications Manager operates in mixed mode.
- Cisco Unified Communications Manager
- Bulk Administration Tool (BAT)
- BAT and the Tool for Auto-Registered Phones Support (TAPS)

For general information about configuring phones in Cisco Unified Communications Manager, refer to the following documentation:

- “Cisco Unified IP Phones”, *Cisco Unified Communications Manager System Guide*
- “Cisco Unified IP Phone Configuration”, *Cisco Unified Communications Manager Administration Guide*
- “Autoregistration Configuration”, *Cisco Unified Communications Manager Administration Guide*

The following section provides additional information.

Set Up Cisco Unified IP Phone 6901 and 6911 in Cisco Unified Communications Manager

The following procedure provides an overview of configuration tasks for the Cisco Unified IP Phone 6901 and 6911 in Cisco Unified Communications Manager Administration. The procedure presents a suggested order to guide you through the phone configuration process. Some tasks are optional, depending on your system and user needs. For detailed procedures and information, see the sources in the procedure.

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
- Associated directory number (DN) to assign to the phone
- Cisco Unified Communications Manager user to associate with the phone

Provides list of configuration requirements for setting up phones.

For more information, see the “Cisco Unified IP Phones” chapter in the *Cisco Unified Communications Manager System Guide* and see [Telephony Features Available for Cisco Unified IP Phone, on page 53](#).

Step 2 Verify that you have sufficient unit licenses for your phone. For more information, see the “License Unit Report” chapter in the *Cisco Communications Manager Administration Guide*.

Step 3 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.

The phone with the default settings gets added to the Cisco Unified Communications Manager database. For more information, see the “Cisco Unified IP Phone Configuration” chapter in the *Cisco Communications Manager Administration Guide*.

For information about Product Specific Configuration fields, see ? 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 “User/Phone Add Configuration” chapter in the *Cisco Communications Manager Administration Guide*.

Step 4 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.

For more information, see the “Directory Number Configuration” chapter in the *Cisco Unified Communications Manager Administration Guide* and see [Telephony Features Available for Cisco Unified IP Phone, on page 53](#).

Step 5 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.

Note Assign a password (for User Options web pages) and PIN (for accessing the Network Menu from the Interactive Voice Response [IVR]).

Adds user information to the global directory for Cisco Unified Communications Manager (Unified CM).

For more information, see the “End User Configuration” chapter in the *Cisco Unified Communications Manager Administration Guide* and see [Add Users to Cisco Unified Communications Manager, on page 62](#).

Note If you want to add both the phone and user to the Cisco Unified Communications Manager database at the same time, see the “User/Phone Add Configurations” chapter in the *Cisco Unified Communications Manager Administration Guide*.

Step 6 Associate a user to a user group. 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.

Note For end users to access Cisco Unified CM User Options, you must add users to the standard Cisco CCM End Users group.

For more information, see the following sections in the *Cisco Unified Communications Manager Administration Guide*:

- End User Configuration Settings section in the “End User Configuration” chapter.
- Adding Users to a User Group section in the “User Group Configuration” chapter.

Step 7 Associate a user with a phone. Provides users with control over their phone for actions such as forwarding calls or adding speed-dial numbers or services.

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

For more information, see the Associating Devices to an End User section in the “End User Configuration” chapter in the *Cisco Unified Communications Manager Administration Guide*.

Cisco Unified IP Phones Installation

After you have added the phones to the Cisco Unified Communications Manager database, you can complete the phone installation. You (or the phone user) can install the phone at the location of the user.



Note Upgrade the phone with the current firmware image before you install the phone. For information about upgrading, see the Readme file for your phone, located at:

<http://tools.cisco.com/support/downloads/go/Redirect.x?mdfid=278875240>

For instructions on upgrading the firmware, see the Release Notes, located at:

http://www.cisco.com/en/US/products/ps10326/prod_release_notes_list.html

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.

Install Cisco Unified IP Phone 6901 and 6911

The following procedure provides an overview of installation tasks for the Cisco Unified IP Phone 6901 and 6911. The procedure presents a suggested order to guide you through the phone installation. Some tasks are optional, depending on your system and user needs. For detailed procedures and information, see the sources in the procedure.

Procedure

- Step 1** Choose the power source for the phone:
- Power over Ethernet (PoE)
 - External power supply
- Determines how the phone receives power. For more information, see [Cisco Unified IP Phone Power, on page 32](#).
- Step 2** Assemble the phone, adjust phone placement, and connect the network cable.
- Locates and installs the phone in the network. See [Install Cisco Unified IP Phone, on page 43](#).
See [Footstand, on page 8](#).
- Step 3** Monitor the phone startup process. Associates directory numbers to the phone and verifies that phone is configured properly.
- See [Phone Startup Verification, on page 45](#).
- Step 4** 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.
- **Using DHCP**
Verify that the phone has DHCP enabled using the IVR. You can set an alternate TFTP server by entering the IP address for the TFTP when prompted by the IVR.
Note Consult with the network administrator to determine whether you need to assign an alternative TFTP server instead of using the TFTP server assigned by DHCP.
 - **Without DHCP**
Verify that the phone has DHCP disabled using the IVR. You must then configure the IP address, subnet mask, TFTP server, and default router locally by using the IVR on the phone.
- For more information, see [Network Settings, on page 45](#) and [Cisco Unified IP Phone Network Settings Setup, on page 47](#).
- Step 5** Set up security on the phone.
- Provides protection against data tampering threats and identity theft of phones.
For more information, see [Cisco Unified IP Phone Security, on page 45](#).
- Step 6** Make calls with the Cisco Unified IP Phone. Verifies that the phone and features work correctly. For more information, see *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.
- Step 7** Provide information to end users about how to use their phones and how to configure their phone options.
- Ensures that users have adequate information to successfully use their Cisco Unified IP Phones.
See [Internal Support Web Site, on page 111](#)
-

Phone Power Reduction

The Cisco Unified IP Phone 6901 and 6911 supports Cisco EnergyWise (EW) (also known as Power Save Plus). When your network contains an EnergyWise controller, you can configure these phones to sleep (power down) and wake (power up) on a schedule to reduce your power consumption.

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

Related Topics

[Phone Configuration Files](#), on page 33

[EnergyWise Setup on Cisco Unified IP Phone 6901 and 6911](#), on page 65

Terminology Differences

The following table highlights some of the important differences in terminology used in these documents:

- *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*
- *Cisco Unified IP Phone 6901 and 6911 Administration Guide for Cisco Unified Communications Manager (SCCP and SIP)*
- *Cisco Unified Communications Manager Administration Guide*
- *Cisco Unified Communications Manager System Guide*

| User Guide | Administration and System Guides |
|--------------------|---|
| Auto Barge | cBarge |
| Message Indicators | Message Waiting Indicator (MWI) or Message Waiting Lamp |
| Voice mail system | Voice messaging system |



CHAPTER 2

Cisco Unified IP Phone and Telephony Networks

- [Phone and Telephony Networks Overview, on page 29](#)
- [Cisco Unified IP Communications Product Interactions, on page 29](#)
- [Cisco Unified IP Phone Power, on page 32](#)
- [Phone Configuration Files, on page 33](#)
- [Phone Startup Process, on page 34](#)
- [Cisco Unified Communications Manager Phone Addition Methods, on page 35](#)
- [Cisco Unified IP Phones and Different Protocols, on page 38](#)
- [Cisco Unified IP Phone MAC Address Determination, on page 40](#)

Phone and Telephony Networks Overview

Cisco Unified IP Phones enable you to communicate by using voice over a data network. To provide this capability, the IP Phones depend upon and interact with several other important Cisco Unified IP Telephony components, including Cisco Unified Communications Manager.

This chapter focuses on the interactions between the Cisco Unified IP Phone 6901 and 6911, Cisco Unified Communications Manager, DNS and DHCP servers, TFTP servers, and switches. It also describes options for powering phones.

For related information about voice and IP communications, go to this URL:

<http://www.cisco.com/c/en/us/products/unified-communications/index.html>

Cisco Unified IP Communications Product Interactions

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

Cisco Unified IP Phone and Cisco Unified Communications Manager Interactions

Cisco Unified Communications Manager is an open and 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 IP telephony system: 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
- Configuration file using the TFTP service
- 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 IP phones described in this chapter, see the “Cisco Unified IP Phone Configuration” chapter in the *Cisco Communications Manager Administration Guide*.



Note If the Cisco Unified IP Phone model that you want to configure does not appear in the Phone Type drop-down list in Cisco Unified Communications Manager, go to the following URL and install the latest support patch for your version of Cisco Unified Communications Manager:

<http://tools.cisco.com/support/downloads/go/Redirect.x?mdfid=278875240>

For more information, see the “Software Upgrades” chapter in the *Cisco Unified Communications Operating System Administration Guide*.

Related Topics

[Cisco Unified IP Phones Security Features](#), on page 16

[Telephony Features Available for Cisco Unified IP Phone](#), on page 53

[Cisco Unified IP Phone and Cisco Unified Communications Manager Express Interaction](#), on page 31

Cisco Unified IP Phone 6911 and VLAN Interaction

The Cisco Unified IP Phone 6911 has an internal Ethernet switch, enabling forwarding of packets to the phone, and to the access port and the network port on the back of the phone.



Note The Cisco Unified IP Phone 6901 does not have a PC port.

If a computer is connected to the 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 connected to the same port.
- Data traffic present on the VLAN supporting phones might reduce the quality of VoIP traffic.
- Network security may require isolation of 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 that the phone connects to would be configured to have 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 connected to the switch through the 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 when there are not enough IP addresses for each phone.

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

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

Related Topics

[Phone Startup Process](#), on page 34

[Cisco Unified IP Phone Network Settings Setup](#), on page 47

Cisco Unified IP Phone and Cisco Unified Communications Manager Express Interaction

When the Cisco Unified IP Phone works with the Cisco Unified Communications Manager Express (Unified CME), the phones 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 Cisco Unified IP 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

Not supported.

Direct Transfer

Not supported.

Select

Not supported.

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

Cisco Unified IP Phone Power

The Cisco Unified IP Phone 6901 and 6911 can be powered with external power or with Power over Ethernet (PoE). A separate power supply provides external power. A switch provides PoE through the Ethernet cable attached to a phone.



Caution

When you install an externally powered phone, connect the power supply to the phone and to a power outlet before you connect the Ethernet cable to the phone. When you remove an externally powered phone, disconnect the Ethernet cable from the phone before you disconnect the power supply.

Power Guidelines

The following table provides guidelines for powering the Cisco Unified IP Phone 6901 and 6911.

Table 6: Guidelines for Powering the Cisco Unified IP Phone 6901 and 6911

| Type | Guidelines |
|---|---|
| External power, provided through the CP-PWR-CUBE-3 external power supply. | The Cisco Unified IP Phone 6901 and 6911 use the CP-PWR-CUBE-3 power supply. |
| External power, provided through the Cisco Unified IP Phone Power Injector. | The Cisco Unified IP Phone Power Injector may be used with any Cisco Unified IP Phone. Functioning as a midspan device, the injector delivers inline power to the attached phone. The Cisco Unified IP Phone Power Injector connects between a switch port and the IP Phone, and supports a maximum cable length of 100 meter (m) between the unpowered switch and the IP Phone. |
| PoE power, provided by a switch through the Ethernet cable attached to the phone. | <p>The Cisco Unified IP Phone 6901 and 6911 support Cisco inline PoE.</p> <p>The Cisco Unified IP Phone 6901 and 6911 support IEEE 802.3af Class 1 power on signal pairs and spare pairs.</p> <p>To ensure uninterruptible operation of the phone, make sure that the switch has a backup power supply.</p> <p>Make sure that the CatOS or IOS version running on your switch supports your intended phone deployment. Refer to the documentation for your switch for operating system version information.</p> |
| External power, provided through inline power patch panel WS-PWR-PANEL | The inline power patch panel WS-PWR-PANEL is compatible with the Cisco Unified IP Phone 6901 and 6911. |

Power Outage

Your access to emergency service through the phone requires the phone to receive power. If an interruption in the power supply occurs, Service and Emergency Calling Service dialing do not function until power is restored. In the case of a power failure or disruption, you may need to reset or reconfigure equipment before you can use the Service or Emergency Calling Service dialing.

Additional Information About Power

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

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

| Document topics | URL |
|---------------------------------------|---|
| Cisco Unified IP Phone Power Injector | http://www.cisco.com/en/US/products/ps6951/index.html |
| PoE Solutions | http://www.cisco.com/en/US/netsol/ns340/ns394/ns147/ns412/index.html |
| Cisco Catalyst Switches | http://www.cisco.com/en/US/products/hw/switches/index.html |
| Integrated Service Routers | http://www.cisco.com/en/US/products/hw/routers/index.html |
| Cisco IOS Software | http://www.cisco.com/en/US/products/ps6951/index.html |

Phone Configuration Files

The TFTP server stores the phone configuration files that define Cisco Unified Communications Manager connection parameters. 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.

In addition, if the device security mode in the configuration file is set to Authenticated and the CTL file on the phone has a valid certificate for Cisco Unified Communications Manager, the phone establishes a TLS connection to Cisco Unified Communications Manager. Otherwise, the phone establishes a TCP connection.



Note If the device security mode in the configuration file is set to Authenticated or Encrypted, but the phone has not received a CTL file, the phone tries four times to obtain a CTL file so it can register securely.

If you configure security-related settings in Cisco Unified Communications Manager, 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 “Configuring Encrypted Phone Configuration Files” chapter in

Cisco Unified Communications Manager Security Guide. 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

If autoregistration is not enabled and the phone is not added to the Cisco Unified Communications Manager database, the system rejects the phone registration request.

If the phone registers and works in encrypted mode, the phone accesses the configuration file named `SEPmac_addr.cnf.xml.enc.sgn`. If the `SEPmac_addr.cnf.xml.enc.sgn` does not exist on the TFTP server, the phone requests the file `SEPmac_addr.cnf.xml.sgn`. That is, if the phone works in encrypted mode with TFTP Encrypted Config selected, the phone accesses the configuration file named `SEPMac_addr.cnf.xml.enc.sgn`. If the phone works in encrypted mode with TFTP Encrypted Config not selected, the phone accesses the file `SEPMac_addr.cnf.xml.sgn`. To enable TFTP Encrypted Configuration, select the TFTP Encrypted Config check box in the Phone Security Profile Configuration page. You can access this page from the Cisco Unified Communications Manager by selecting **System > Security > Phone Security Profile** and clicking **Add New**.

For SIP phones, the TFTP server generates these SIP configuration files:

- SIP IP Phone:
 - For unsigned and unencrypted files - `SEP<mac>.cnf.xml`
 - For signed files - `SEP<mac>.cnf.xml.sgn`
 - For signed and encrypted files - `SEP<mac>.cnf.xml.enc.sgn`
- Dial Plan - `<dialplan>.xml`

The filenames derive from the MAC address and description fields in the Phone Configuration window of Cisco Unified Communications Manager. The MAC address uniquely identifies the phone.

For more information on phone configuration settings, see the “Cisco Unified IP Phone Configuration” chapter in the *Cisco Communications Manager Administration Guide*.

Phone Startup Process

When connecting to the VoIP network, the Cisco Unified IP Phone 6901 and 6911 go through a standard startup process that is described in the following list. Depending on your specific network configuration, not all these steps may occur on your Cisco Unified IP Phone.

1. Obtain power from the switch. If a phone does not use external power, the switch provides in-line power through the Ethernet cable attached to the phone. For more information, see [Cisco Unified Communications Manager Phone Addition Methods, on page 35](#) and [Startup Problems, on page 93](#).
2. Configure VLAN. If the Cisco Unified IP Phone connects to a Cisco Catalyst switch, the switch informs the phone of the voice VLAN defined on the switch. The phone needs to know the VLAN membership before proceeding with the Dynamic Host Configuration Protocol (DHCP) request for an IP address.

For more information, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#) and [Startup Problems, on page 93](#).

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

For more information, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#) and [Startup Problems, on page 93](#).

4. Access a TFTP server. In addition to assigning an IP address, the DHCP server directs the Cisco Unified 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 alternative TFTP server to use instead of the one assigned by DHCP.

For more information, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#) and [Startup Problems, on page 93](#).

5. Request the configuration file. The TFTP server stores configuration files that define parameters for connecting to Cisco Unified Communications Manager and other information for the phone.

For more information, see [Cisco Unified Communications Manager Phone Addition Methods, on page 35](#) and [Startup Problems, on page 93](#).

6. Load the stored phone image. The Cisco Unified IP Phone has nonvolatile flash memory to store firmware images and user-defined preferences. At startup, the phone runs a bootstrap loader that loads a phone image stored in flash memory. Using this image, the phone initializes the software and hardware.

For more information, see [Startup Problems, on page 93](#).

7. Contact Cisco Unified Communications Manager. The configuration file defines how the Cisco Unified IP Phone communicates with Cisco Unified Communications Manager and provides a phone with the load ID. After obtaining 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 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 in the Cisco Unified Communications Manager database.

For more information, see [Startup Problems, on page 93](#).

Cisco Unified Communications Manager Phone Addition Methods

Before installing the Cisco Unified IP Phone, you must choose a method for adding phones to the Cisco Unified Communications Manager database. Each phone type requires a fixed number of device license units and the available number of unit licenses on the server may impact phone registration. For more information on licensing, see the “Licenses for Phones” section in the *Cisco Unified Communications Manager System Guide*.

The following table provides an overview of the Cisco Unified Communications Manager phone addition methods.

Table 7: Methods for Adding Phones to the Cisco Unified Communications Manager Database

| Method | Requires MAC address? | Notes |
|---|-----------------------|--|
| Autoregistration | No | Results in automatic assignment of directory numbers. Not available when security or encryption is enabled. |
| Autoregistration with TAPS | No | Requires autoregistration and the Bulk Administration Tool (BAT); updates information in the Cisco Unified IP Phone and in Cisco Unified Communications Manager Administration |
| Using the Cisco Unified Communications Manager Administration | Yes | Requires phones to be added individually |
| Using BAT | Yes | Allows for simultaneous registration of multiple phones |

Autoregistration Phone Addition

By enabling autoregistration before you begin installing phones, you can:

- Add phones without first gathering MAC addresses from the phones.
- Automatically add a Cisco Unified 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.



Note Cisco recommends you use autoregistration to add less than 100 phones to your network. To add more than 100 phones to your network, use the Bulk Administration Tool (BAT). See [Add Phones Using BAT Phone Template, on page 38](#).

The system disables autoregistration 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. For information about enabling

autoregistration, see the “Enabling Autoregistration” section in the *Cisco Unified Communications Manager Administration Guide*.



Note When you configure the cluster for mixed mode through the Cisco CTL client, autoregistration automatically disables. When you configure the cluster for nonsecure mode through the Cisco CTL client, autoregistration automatically enables.

Autoregistration and TAPS Phone Addition

You can add phones with autoregistration the Tool for Auto-Registered Phones Support (TAPS), 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 download predefined phone configuration files.



Note Cisco recommends you use autoregistration and TAPS to add less than 100 phones to your network. To add more than 100 phones to your network, use the Bulk Administration Tool (BAT). See [Add Phones Using BAT Phone Template, on page 38](#).

To implement TAPS, you or the end user dial a TAPS directory number and follow the voice prompts. When the process completes, the phone has downloaded the directory number and other settings, and the Cisco Unified Communications Manager database contains the correct MAC address for the phone.

Autoregistration must be enabled in Cisco Unified Communications Manager (**System > Cisco Unified CM**) for TAPS to function.



Note When you configure the cluster for mixed mode through the Cisco CTL client, autoregistration is automatically disabled. When you configure the cluster for nonsecure mode through the Cisco CTL client, autoregistration is automatically enabled.

For more information, see the “Bulk Administration” chapter in *Cisco Unified Communications Manager Administration Guide*.

Cisco Unified Communications Manager Phone Addition

You can add phones individually to the Cisco Unified Communications Manager database using Cisco Unified Communications Manager administration. To do so, you first obtain the MAC address for each phone.

For information about determining a MAC address, see [Cisco Unified IP Phone MAC Address Determination, on page 40](#).

After you have collected MAC addresses, in Cisco Unified Communications Manager Administration, choose **Device > Phone** and click **Add New** to begin.

For complete instructions and conceptual information about Cisco Unified Communications Manager, see the “Cisco Unified Communications Manager Overview” chapter in the *Cisco Unified Communications Manager System Guide*.

Add Phones Using BAT Phone Template

The Cisco Unified Communications Manager Bulk Administration Tool (BAT) enables you to perform batch operations, including registration, on multiple phones. To access BAT, choose Bulk Administration in Cisco Unified Communications Manager Administration,

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

For information about determining a MAC address, see [Cisco Unified IP Phone MAC Address Determination, on page 40](#).

For detailed instructions about adding phones using the Bulk Administration menu, see the *Cisco Unified Communications Manager Bulk Administration Guide*, chapter “Inserting Phones”.

To add a phone to the Cisco Unified Communications Manager using the BAT phone template, perform these steps:

Procedure

- Step 1** From Cisco Unified Communications Manager, 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, including Device Pool, Phone Button Template, and Device Security Profile.
 - Step 5** Click **Save**.
 - Step 6** From Cisco Unified Communications Manager, choose **Device > Phone > Add New** to add a phone using an existing BAT phone template.
-

Cisco Unified IP Phones and Different Protocols

The Cisco Unified IP Phone can operate with SCCP (Skinny Client Control Protocol) or SIP (Session Initiation Protocol). You can convert a phone that is using one protocol to use the other protocol.

Convert New Phone from SCCP to SIP

A new, unused phone uses SCCP. To convert this phone to SIP, perform these steps:

Procedure

- Step 1** Take one of these actions:

- To autoregister the phone, set the Auto Registration Phone Protocol parameter in Cisco Unified Communications Manager to SIP.
- To provision the phone using the Bulk Administration Tool (BAT), choose the appropriate phone model and choose SIP from the BAT.

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

- To provision the phone manually, make the appropriate changes for SIP on the Phone configuration window in Cisco Unified Communications Manager.

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

- Step 2** If you are not using DHCP in your network, configure the network parameters for the phone. See [Network Settings, on page 45](#).
- Step 3** To save the configuration updates,
- a) Click Apply Config.
 - b) When the Apply Configuration Information window displays, click OK.
 - c) Tell the user to power cycle the phone.
-

In-Use Phone Protocol to Protocol Conversion

Phones using SCCP can be upgraded to use SIP. To change from SCCP to SIP, the phone firmware must be updated to the recommended SIP version before the phones can register. New Cisco Unified IP Phones ship from the factory with SCCP phone firmware. These new phones must be upgraded to the recommended SIP version before they can complete registration.

For information about how to convert an in-use phone from one protocol to the other, see the *Cisco Unified Communications Manager Administration Guide*, chapter “Cisco Unified IP Phone Configuration”, section “Migration Existing Phone Configuration to a Different Phone”.

Deploy Phone in SCCP and SIP Environment

To deploy Cisco Unified IP Phones in an environment that includes SCCP and SIP and in which the Cisco Unified Communications Manager autoregistration parameter specifies SCCP, perform these general steps:

Procedure

- Step 1** Set the Cisco Unified Communications Manager `auto_registration_protocol` parameter to SCCP.
- Step 2** From Cisco Unified Communications Manager, choose **System > Enterprise Parameters**.
- Step 3** Install the phones.
- Step 4** Change the Auto Registration Protocol enterprise parameter to SIP.
- Step 5** Autoregister the SIP phones.
-

Cisco Unified IP Phone MAC Address Determination

Several procedures described in this manual require you to determine the MAC address of a Cisco Unified IP Phone. You can determine the MAC address of a phone in these ways:

- Look at the MAC label on the back of the phone.
- Display the web page for the phone and click the **Device Information** hyperlink.

Related Topics

[Access Web Page for Phone](#), on page 74



CHAPTER 3

Cisco Unified IP Phone Installation

- [Phone Installation Overview](#), on page 41
- [Before You Begin](#), on page 41
- [Cisco Unified IP Phone 6901 and 6911 Components](#), on page 42
- [Install Cisco Unified IP Phone](#), on page 43
- [Phone Startup Verification](#), on page 45
- [Network Settings](#), on page 45
- [Cisco Unified IP Phone Security](#), on page 45

Phone Installation Overview

This chapter helps you install the Cisco Unified IP Phone on an IP telephony network.



Note Before you install a Cisco Unified IP Phone, you must decide how to configure the phone in your network. Then you can install the phone and verify the functionality. For more information, see [Cisco Unified IP Phone and Telephony Networks](#), on page 29.

Before You Begin

Before installing the Cisco Unified IP Phone, review the requirements in the following sections.

Network Requirements

For the Cisco Unified IP Phone to successfully operate as a Cisco Unified IP Phone endpoint in your network, your network must meet the following requirements:

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

- IP network that supports DHCP or manual assignment of IP address, gateway, and subnet mask



Note The Cisco Unified IP 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. If the Cisco Unified Communications Manager server is located in a different time zone than the phones, the phones do not display the correct local time.

Cisco Unified Communications Manager Setup

The Cisco Unified IP Phone requires Cisco Unified Communications Manager to handle call processing. See *Cisco Unified Communications Manager Administration Guide* or refer to the context-sensitive help in the Cisco Unified Communications Manager application to ensure that Cisco Unified Communications Manager is set up properly to manage the phone and to properly route and process calls.

If you plan to use autoregistration, verify that it is enabled and properly configured in Cisco Unified Communications Manager before connecting any Cisco Unified IP Phone to the network. For information about enabling and configuring autoregistration, see *Cisco Unified Communications Manager Administration Guide*.

You must use Cisco Unified Communications Manager Administration to configure and assign telephony features to the Cisco Unified IP Phones.

In Cisco Unified Communications Manager Administration, you can add users to the database, add users to user groups, and associate users to specific phones. In this way, users gain access to their Cisco Unified CM User Option page to configure items such as call forwarding, speed dialing, and voice messaging system options.

Related Topics

- [Cisco Unified Communications Manager Phone Addition Methods](#), on page 35
- [Telephony Features Available for Cisco Unified IP Phone](#), on page 53
- [Add Users to Cisco Unified Communications Manager](#), on page 62

Cisco Unified IP Phone 6901 and 6911 Components

The following sections describe the components and accessories for the Cisco Unified IP Phone 6901 and 6911.

Network and Access Ports

The back of the Cisco Unified IP Phone 6901 includes a network port, which the phone uses to connect to the network.

The back of the Cisco Unified IP Phone 6911 includes these ports:

- Network port: Labeled Network
- Access port: Labeled Computer (also known as the PC port)



Note Only the Cisco Unified IP Phone 6911 has a PC port. The Cisco Unified IP Phone 6901 does not have a PC port.

The network port supports 10/100 megabits per second (Mb/s) half- or full-duplex connections to external devices. You can use either Category 3, 5, or 5e cabling for 10 Mbps connections, but you must use Category 5 or 5e for 100 Mb/s connections.

Use the network port to connect the phone to the network. You must use a straight-through cable on this port. The phone can also obtain inline power from a switch over this connection. For more information, see [Cisco Unified Communications Manager Phone Addition Methods, on page 35](#).

Use the PC access port to connect a network device, such as a computer, to the phone. You must use a straight-through cable on this port.

Handset

The Cisco Unified IP Phone uses a handset that is designed especially for the phone. The handset includes a light strip to indicate incoming calls and voice messages waiting.

To connect a handset to the Cisco Unified IP Phone, plug the cable into the handset and into the Handset port on the back of the phone.

Disable Speakerphone (Cisco Unified IP Phone 6911 Only)

By default, the speakerphone is enabled on the Cisco Unified IP Phone.

You can disable the speakerphone using Cisco Unified Communications Manager.

Procedure

- Step 1** Select **Device > Phone**.
 - Step 2** Select the phone you want to modify.
 - Step 3** In the Phone Configuration window for the phone, check the Disable Speakerphone check box.
-

Install Cisco Unified IP Phone

You must connect the Cisco Unified IP Phone to the network and to a power source before using it.

To install a Cisco Unified IP Phone, perform the following steps.

Procedure

- Step 1** Connect the handset to the Handset port.

- Step 2** Connect the power supply to the Cisco DC Adapter port. For guidelines, see [Cisco Unified IP Phone Power, on page 32](#) for guidelines.
- Step 3** Connect a straight-through Ethernet cable from the switch to the network port labeled Network on the Cisco Unified IP Phone 6901 and 6911.
- Each Cisco Unified IP Phone ships with one Ethernet cable in the box.
- You can use either Category 3, 5, or 5e cabling for 10 Mbps connections, but you must use Category 5 or 5e for 100 Mbps connections.
- For guidelines, see the [Network and Access Ports, on page 42](#).
- Step 4** (Cisco Unified IP Phone 6911 only) Connect a straight-through Ethernet cable from another network device, such as a desktop computer, to the access port labeled Computer.
- You can connect another network device later if you do not connect one now.
- You can use either Category 3, 5, or 5e cabling for 10 Mbps connections, but you must use Category 5 or 5e for 100 Mbps connections.
- For guidelines, see the [Network and Access Ports, on page 42](#).

Related Topics

- [Footstand](#), on page 8
- [Phone Startup Verification](#), on page 45
- [Network Settings](#), on page 45

Cisco Unified IP Phone 6901 Installation

Use the diagram and table in [Phone Connections, on page 2](#) to attach cables to the phone.

Cisco Unified IP Phone 6911 Installation

Use the diagram and table in [Phone Connections, on page 4](#) to attach cables to the phone.

Phone Wall Mount

You can mount the Cisco Unified IP Phone 6901 on the wall using a standard telephone wall plate with an opening for an RJ-45 connector. Cisco recommends that you use Leviton Wall Mount plate (Leviton type number: 4108W-0SP) to wall mount the Cisco Unified IP Phone 6901.

You can mount the Cisco Unified IP Phone 6911 on the wall using special brackets available in the Cisco Unified IP Phone wall mount kit.



Note The phone does not include wall mount kits. Wall mount kits must be ordered separately.

Related Topics

- [Cisco Unified IP Phone 6911 Installation](#), on page 44
- [Cisco Unified IP Phone Wall Mount](#), on page 129

[Cisco Unified IP Phone Non-Lockable Wall Mount](#), on page 139

Phone Startup Verification

The message waiting (MWI) LED on the handset and the Mute button LED light during boot up. After successful completion of the boot up process, the LEDs turn off, and the phone is ready for calls. If the phone successfully boots up, it has started up properly. If the phone does not start up properly, the user receives a reorder tone.

Related Topics

[Startup Problems](#), on page 93

Network Settings

If you are not using DHCP in your network, you must configure the Cisco Unified IP Phone after installing the phone on the network. You configure the following network settings:

- IP address (IPv4)
- IP subnet information
- IPv6 address: you only configure the IPv6 address if the phone is in an IPv6 network. Other IPv6 parameters include the IPv6 prefix, IPv6 default gateway, and IPv6 TFTP server.
- Default Router
- TFTP server IP address

Collect this information and see the instructions in [Cisco Unified IP Phone Settings](#), on page 47.

Cisco Unified IP Phone Security

The 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 digitally sign files before they are delivered.

For more information about the security features, see the related topics and the *Cisco Unified Communications Manager Security Guide*.

Locally Significant Certificate Manual Setup

You initiate the installation of a locally significant certificate (LSC) from the Security Configuration menu on the phone. This menu also lets you update or remove an LSC.

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 file should have a CAPF certificate.

- On Cisco Unified Communications Operating System Administration, verify that the CAPF certificate has been installed.
- The CAPF is running and configured.

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



CHAPTER 4

Cisco Unified IP Phone Settings

- [Phone Settings Overview](#), on page 47
- [Cisco Unified IP Phone Network Settings Setup](#), on page 47
- [Access Phone Configuration Settings](#), on page 48

Phone Settings Overview

The Cisco Unified IP Phone includes configurable network settings that you may need to modify before the phone can be used. You can access, view, and change the network settings on the phone by accessing the Interactive Voice Response (IVR) tool. Other configuration settings may be configured in Cisco Unified Communications Manager.

Cisco Unified IP Phone Network Settings Setup

Before the Cisco Unified IP Phone can function, you must configure the network setting on the phone. You can review the configuration of a phone by accessing IVR on the phone. When the phone is connected to a network with a DHCP server, you can use the DHCP server to configure the network settings. You can also manually configure the phone by accessing the IVR.

Phone Settings Options

The following table describes the options that can be configured on the IVR.

Table 8: Phone Setting Options

| Option | Description |
|--------|---|
| DHCP | Indicates whether the phone has DHCP enabled or disabled. When DHCP is enabled, the DHCP server assigns to the phone an IP address, subnet mask, default router, and TFTP Server. When DHCP is disabled, the administrator must manually assign the IP address, subnet mask, default router, and TFTP server on the phone. |

| Option | Description |
|------------------------------|--|
| IP Address | Internet Protocol version 4 (IPv4) 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 1 | Default router used by the phone (Default Router 1). |
| TFTP Server | Primary Trivial File Transfer Protocol (TFTP) server used by the phone. If you are not using DHCP in your network or you want to change this server, you must assign the TFTP Server. |
| 802.1 xSecurity | Indicates whether the phone has 802.1x security enabled or disabled. |
| IPv6 Network Settings | |
| DHCPv6 | Dynamic Host Configuration Protocol (DHCP) automatically assigns IPv6 addresses to phones when you connect them to the network. Cisco Unified IP Phones enable DHCP by default. |
| IPv6 Default Router 1 | Default IPv6 router used by the phone (Default Router 1). |
| IPv6 Address | IPv6 address of the phone. The IPv6 address is a 128 bit address. |
| IPv6 Prefix Length | Subnet prefix length used by the phone. The subnet prefix length is a decimal value from 1 to 128 that specifies the portion of the IPv6 address that comprises the subnet. |
| IPv6 TFTP Server | Indicates whether the phone uses the IPv6 Trivial File Transfer Protocol (TFTP) server. |

Access Phone Configuration Settings

You can change the PIN of the IVR from the Common Phone Profile page.



Note When there is no User ID or PIN associated with a phone, the phone uses a default PIN of 24726.

Procedure

- Step 1** Access the Cisco Unified Communications Manager administration.
- Step 2** Navigate to **Device > Device Settings > Common Phone Profile**.
- Step 3** Specify a PIN for the Local Phone Unlock PIN field in the Common Phone Profile Configuration window.
- Step 4** Click **Save**.
-

Related Topics

[Cisco Unified IP Phone Network Settings Setup](#), on page 47

Access IVR and Set Up Phone Settings

To access the IVR and configure the phone settings, follow these steps on the phone:



Note You can change the PIN by using the Local Phone Unlock Password in the Common Phone Profile Configuration page of the Cisco Unified CM Administration web page.

Procedure

- Step 1** Go off-hook and press star (*), pound (#), and 0 keys simultaneously. On the Cisco Unified IP Phone 6911, you can also access IVR by pressing the speaker button followed by the *, #, and 0 keys simultaneously. The IVR prompts for a password.
- Note** The Cisco Unified IP Phone 6901 and 6911 supports alphanumeric characters (A-F), and colon (:) in IPv6 settings.
- Step 2** Enter the PIN using the keypad, followed by #.
You are at the IVR main configuration menu.
- Step 3** Follow the voice prompts on the IVR. For more information on navigating the IVR, see [IVR Configuration Menu, on page 49](#).
- Step 4** To return to the main configuration menu, press #.
- Step 5** To exit the IVR, end the call.
-

IVR Configuration Menu

When you input information in response to IVR prompts, use the following list for special character input:

- To enter a period (.) or colon (:) that separates octets in the IP Address, press star (*).
- To enter the hexadecimal A, press the 2 key two times quickly.
- To enter the hexadecimal B, press the 2 key three times quickly

- To enter the hexadecimal C, press the 2 key four times quickly.
- To enter the hexadecimal D, press the 3 key two times quickly.
- To enter the hexadecimal E, press the 3 key three times quickly.
- To enter the hexadecimal F, press the 3 key four times quickly.
- To delete a character in an entry, press Redial.

The following table describes the options in the IVR Configuration menu.

Table 9: Navigating the IVR Configuration Menu

| Action | IVR Code | Navigating Notes |
|--------------------------------|----------|---|
| Review or Set Network Settings | 1 | <p>If DHCP is enabled, the IVR announces each network parameter.</p> <p>If DHCP is disabled, the IVR announces each network parameter, pausing between each parameter to allow you to enter a new value.</p> <ol style="list-style-type: none"> 1. IVR announces the IP Address. To change the IP address for the phone, use the keypad to enter a new IP address, followed by #. To retain the current IP address, press #. 2. IVR announces the subnet mask. To change the subnet mask, use the keypad to enter a new IP address, followed by #. To retain the current subnet mask, press #. 3. IVR announces the default gateway. To change the default gateway, use the keypad to enter a new IP address, followed by #. To retain the current default gateway, press #. |

| Action | IVR Code | Navigating Notes |
|-------------------------------------|----------|--|
| Review or Set TFTP Server | 2 | <p>IVR announces the current TFTP setting.</p> <p>To change the TFTP server, use the keypad to enter a new address, followed by #. To retain the current TFTP server, press #.</p> <p>To reset a manually-configured TFTP server to the DHCP-configured server, press *.</p> |
| Enable or Disable DHCP | 3 | Press 3 to enable or disable DHCP. |
| Enable or Disable 802.1X | 4 | Press 4 to enable or disable 802.1X security. |
| Review or Set IPv6 Network Settings | 5 | <p>If DHCPv6 is enabled, the IVR announces each network parameter sequentially: IPv6 address, IPv6 prefix length, IPv6 subnet mask, and IPv6 default gateway.</p> <p>If DHCPv6 is disabled, the IVR announces each network parameter, pausing between each parameter to allow you to enter a new value.</p> <ol style="list-style-type: none"> 1. IVR announces the IPv6 Address. To change the IPv6 address for the phone, use the keypad to enter a new IPv6 address, followed by #. To retain the current IPv6 address, press #. 2. IVR announces the subnet mask. To change the subnet mask, use the keypad to enter a new IPv6 address, followed by #. To retain the current subnet mask, press #. 3. IVR announces the default gateway. To change the default gateway, use the keypad to enter a new IPv6 address, followed by #. To retain the current default gateway, press #. |

| Action | IVR Code | Navigating Notes |
|--------------------------------|----------|--|
| Review or Set Ipv6 TFTP Server | 6 | <p>IVR announces the current IPv6 TFTP setting.</p> <p>To change the IPv6 TFTP server, use the keypad to enter a new IPv6 address, followed by #. To retain the current IPv6 TFTP server, press #.</p> <p>To reset a manually-configured IPv6 TFTP server to the IPv6 DHCP-configured server, press *.</p> |
| Enable or Disable DHCPv6 | 7 | Press 7 to enable or disable DHCPv6. |
| Reset to factory settings | 0 | <p>Note All configuration settings change to the default factory settings and the phone resets. It takes some time for the phone to reregister. Select this option only when needed.</p> |



CHAPTER 5

Features, Templates, Services, and User Setup

- [Features, Templates, Services, and Users Overview](#), on page 53
- [Telephony Features Available for Cisco Unified IP Phone](#), on page 53
- [Add Users to Cisco Unified Communications Manager](#), on page 62
- [User Options Web Pages Management](#), on page 63
- [EnergyWise Setup on Cisco Unified IP Phone 6901 and 6911](#), on page 65
- [UCR 2008 Setup](#), on page 68

Features, Templates, Services, and Users Overview

After you install Cisco Unified IP Phones in your network, configure their network settings, and add them to Cisco Unified Communications Manager, you use the Cisco Unified Communications Manager Administration application to configure telephony features, modify phone templates, set up services, and assign users.

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

To list supported features for all phones or for a particular phone model on your Cisco Unified Communications Manager (Unified CM), you can generate a Unified CM Phone Feature List report with Cisco Unified Reporting.

For suggestions about how to provide users with information about features, and what information to provide, see [Internal Support Web Site](#), on page 111.

For information about setting up phones in non-English environments, see [International User Support](#), on page 113.

Telephony Features Available for Cisco Unified IP Phone

After you add Cisco Unified 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 using Cisco Unified Communications Manager Administration. The Reference column lists Cisco Unified Communications Manager, other documentation, and links to sections in this document that contains configuration procedures and related information.

For information about using most of these features on the phone, see *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.



Note Cisco Unified Communications Manager also provides several service parameters that you can use to configure various telephony functions. For more information on accessing and configuring service parameters, see *Cisco Unified Communications Manager Administration Guide*.



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

Table 10: Telephony Features for the Cisco Unified IP Phones

| Feature | Description | Configuration reference |
|--|---|---|
| Agent Greeting | <p>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 record greetings and update them, as required.</p> <p>When a customer calls, both the agent and the customer can hear the prerecorded greeting. The agent can remain on mute until the greeting ends or answer the call over the greeting.</p> <p>All codecs supported for the phone are supported for Agent Greeting calls.</p> <p>To enable Agent Greeting in the Cisco Unified CM Administration application, choose Device > Phone, and locate the IP Phone that you want to configure. Scroll to the Device Information Layout pane and set the Built In Bridge field to On or Default.</p> <p>If Built In Bridge is set to Default, choose System > Service Parameter and select the appropriate Server and Service. Scroll to the Clusterwide Parameters (Device - Phone) pane and set Builtin Bridge Enable to On.</p> | <p>For more information, see:</p> <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Features and Services Guide</i>, “Barge and Privacy” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phones” chapter |
| Audible Message Waiting Indicator (AMWI) | A stutter tone from the handset or speakerphone indicates that a user has one or more new voice messages on a line. | For more information, see the <i>Cisco Unified Communications Manager System Guide</i> , “Cisco Unified IP Phone” chapter. |

| Feature | Description | Configuration reference |
|--|---|--|
| Auto Answer (Cisco Unified IP Phone 6911 only) | Connects incoming calls automatically after a ring or two. Auto Answer works with the speakerphone. | For more information, see the <i>Cisco Unified Communications Manager Administration Guide</i> , “Directory Number Configuration” chapter. |
| Automatic Port Synchronization (Cisco Unified IP Phone 6911 only) | <p>When the Cisco Unified CM administrator uses the Remote Port Configuration feature to set the speed and duplex function of an IP Phone remotely, loss of packets can occur if one port is slower than the other.</p> <p>The Automatic Port Synchronization feature synchronizes the ports to the lowest speed among the two ports to eliminate packet loss. When automatic port synchronization is enabled, Cisco recommends that both ports autonegotiate. If one port autonegotiates and the other uses a fixed speed, the phone synchronizes to the fixed port speed.</p> <p>Note If both the ports use fixed speed, the Automatic Port Synchronization feature is ineffective. The Remote Port Configuration and Automatic Port Synchronization features are compatible only with IEEE 802.3AF Power of Ethernet (PoE) switches. Switches that support only Cisco Inline Power are not compatible. Enabling this feature on phones that are connected to these types of switches could result in loss of connectivity to Cisco Unified CM if the phone is powered by PoE.</p> | <p>To configure the parameter in the Cisco Unified CM Administration application, choose Device > Phone, select the appropriate IP phones, and scroll to the Product Specific Configuration Layout pane.</p> <p>To configure the setting on multiple phones simultaneously, enable Automatic Port Synchronization in one of the following windows:</p> <ul style="list-style-type: none"> • Enterprise Phone Configuration (System > Enterprise Phone Configuration) • Common Phone Profile Configuration (Device > Device Settings > Common Phone Profile) |
| Block External to External Transfer | Prevents users from transferring an external call to another external number. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “External Call Transfer Restrictions” chapter. |

| Feature | Description | Configuration reference |
|---|---|--|
| Call Forward | <p>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.</p> <p>You hear a stutter tone when off hook if Call Forward All is active on your phone.</p> | <p>For more information, see:</p> <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, “Directory Number Configuration” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phone” chapter • Customize User Options Web Pages Options, on page 64 |
| Call Forward All Loop Breakout | <p>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.</p> | <p>For more information, see the <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phone” chapter.</p> |
| Call Forward All Loop Prevention | <p>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.</p> | <p>For more information, see the <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phone” chapter.</p> |
| Call Forward Destination Override | <p>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.</p> | <p>For more information, see the <i>Cisco Unified Communications Manager System Guide</i>, “Understanding Directory Numbers” chapter.</p> |
| Call Pickup (Cisco Unified IP Phone 6911 only) | <p>Allows users to answer a call ringing on a coworker's phone.</p> <p>You can configure an audio alert for the primary line on the phone. This alert notifies the users that a call is ringing in their pickup group.</p> | <p>For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i>, “Call Pickup” chapter.</p> |
| Call Waiting | <p>Indicates and allows users to answer an incoming call while active on another call.</p> <p>The phone sounds the call waiting tone (single beep) and the line button flashes amber.</p> | <p>For more information, see the <i>Cisco Unified Communications Manager System Guide</i>, “Understanding Directory Numbers” chapter.</p> |

| Feature | Description | Configuration reference |
|--|--|---|
| cBarge | Allows a user to join a nonprivate call on a shared phone line. cBarge adds a user to an existing call and converts it into a conference, allowing the user and other parties to access conference features. | For more information, see: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Features and Services Guide</i>, “Barge and Privacy” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phone” chapter • <i>Cisco Unified Communications Manager Features and Services Guide</i>, “Barge and Privacy” chapter |
| Cisco Unified Communications Manager Express (Unified CME) Version Negotiation | The Cisco Unified Communication Manager Express uses a special tag in the information sent to the phone to identify itself. This tag enables the phone to provide services to the user that the switch supports. | For more information, see: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Express System Administrator Guide</i> • Cisco Unified IP Phone and Cisco Unified Communications Manager Express Interaction, on page 31 |
| Cisco Unified Video Advantage (CUVA) (Cisco Unified IP Phone 6911 only) | Allows users to make video calls by using their Cisco Unified IP Phones, personal computers, and external video cameras. | Configure the Video Capabilities and Auto Line Select parameter in the Product Specific Configuration Layout section in Phone Configuration. |
| Cisco Web Dialer | Allows users to make calls from web and desktop applications. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Cisco Web Dialer” chapter. |
| Client Matter Code (CMC) | Enables a user to specify that a call relates to a specific client matter. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Client Matter Codes and Forced Authorization Codes” chapter. |

| Feature | Description | Configuration reference |
|---|---|---|
| Conference | <p>Allows a user to talk simultaneously with multiple parties by calling each participant individually. Conference features include Conference and Meet Me.</p> <p>Allows a participant (host, initiator, or noninitiator) in a standard (ad hoc) conference to add participants; also allows any conference participant to join together two standard conferences on the same line.</p> <p>Allows a conference host using a Cisco Unified IP Phone 6901 to remove the last participant that joined the conference using the hookflash feature.</p> | <p>The Advance Adhoc Conference service parameter, disabled by default in Cisco Unified Communications Manager, allows you to enable these features.</p> <p>For information on conferences, see the <i>Cisco Unified Communications Manager System Guide</i>, “Conference Bridges” chapter.</p> <p>For more information, see the <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phone” chapter.</p> <p>Note Be sure to inform your users if these features are activated.</p> |
| CTI Applications | A computer telephony integration (CTI) route point can designate a virtual device to receive multiple, simultaneous calls for application-controlled redirection. | For more information, see the <i>Cisco Unified Communications Manager Administration Guide</i> , “CTI Route Point Configuration” chapter. |
| EnergyWise | Enables an IP Phone to sleep (power down) and wake (power up) at predetermined times to promote energy savings. | For more information, see EnergyWise Setup on Cisco Unified IP Phone 6901 and 6911 , on page 65. |
| Forced Authorization Codes (FAC) | Controls the types of calls that certain users can place. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Client Matter Codes and Forced Authorization Codes” chapter. |
| Group Call Pickup (Cisco Unified IP Phone 6911 only) | Allows a user to answer a call that is ringing on a phone in another group. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Call Pickup” chapter. |
| Hold Status | <p>Enables phones with a shared line to distinguish between the local and remote lines that placed a call on hold.</p> <p>The Line LED lights green for Local Hold and red for Remote Hold.</p> | No configuration required. |
| Hold/Resume | <p>Allows the user to move a connected call from an active state to a held state by using the Hold button. The user resumes a held call by pressing the line button, speaker button, or going off hook.</p> <p>Note The LED on the line button pulses green when a local call is on hold and the LED pulses red when a remote call is on hold.</p> | No configuration required unless you want to use Music on Hold. For more information, see “Music on Hold” in this table. |

| Feature | Description | Configuration reference |
|---|--|---|
| Jitter Buffer | Handles jitter from 10 milliseconds (ms) to 1000 ms for both audio and video streams. | No configuration required. |
| Meet Me Conference (Cisco Unified IP Phone 6911 only) | Allows a user to host a Meet Me conference in which other participants call a predetermined number at a scheduled time. The user joins a Meet Me conference by pressing the feature button and the Meet Me conference number. | For more information, see the <i>Cisco Unified Communications Manager Administration Guide</i> , “Meet Me Number/Pattern Configuration” chapter . |
| Message Waiting | Defines directory numbers for the message waiting on and message waiting off indicator. A directly connected voice message system uses the specified directory number to set or to clear a message waiting indicator for a particular Cisco Unified IP Phone. | For more information, see: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, “Message Waiting Configuration” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Voice Mail Connectivity to Cisco Unified Communications Manager” chapter |
| Message Waiting Indicator | A light on the handset that indicates that a user has one or more new voice messages. | For more information, see: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, “Message Waiting Configuration” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Voice Mail Connectivity to Cisco Unified Communications Manager” chapter |
| Monitoring and Recording (Cisco Unified IP Phone 6911 only) | Allows a supervisor to monitor an active call silently. The supervisor cannot be heard by either party on the call. The user may receive an audible alert during a call when it is being monitored. 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 the call is secure and is being monitored. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Monitoring and Recording” chapter. |
| Multilevel Precedence and Preemption (MLPP) (SCCP phones only) | Provides a method of prioritizing calls within your phone system. Use this feature when users work in an environment where they need to make and receive urgent or critical calls. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Multilevel Precedence and Preemption” chapter. |

| Feature | Description | Configuration reference |
|--|--|---|
| Music on Hold | Plays music while callers are on hold. | For more information, see the <i>Cisco Unified Communications Manager Features and Services Guide</i> , “Music On Hold” chapter. |
| Mute (Cisco Unified IP Phone 6911 only) | Mutes the microphone on the handset or speakerphone. | No configuration required. |
| On-hook Call Transfer | Allows a user to press the Transfer button and go on-hook to complete a call transfer. | For more information, see the <i>Cisco Unified Communications Manager System Guide</i> , “Cisco Unified IP Phones” chapter. |
| 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 the feature in the Cisco Unified IP Phone 6901 and 6911 only works with off-hook dialing. | No configuration required. |
| Private Line Automated Ringdown (PLAR) | The Cisco Unified Communications Manager administrator can configure a phone number that the Cisco Unified 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. | For more information, see the <i>Cisco Unified Communications Manager Administration Guide</i> , “Directory Number Configuration” chapter. |
| Redial | Allows users to call the most recently dialed phone number by pressing the Redial button. | No configuration required. |
| Remote Port Configuration | Allows the administrator to configure the speed and duplex function of the phone Ethernet ports remotely by using Cisco Unified CM 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 CM, the data cannot be changed on the phone. | To configure the parameter in the Cisco Unified CM Administration application, choose Device > Phone , select the appropriate IP Phone, and scroll to the Product Specific Configuration Layout pane (Switch Port Remote Configuration or PC Port Remote Configuration). To configure the setting on multiple phones simultaneously, configure the remote configuration in either Enterprise Phone Configuration (System > Enterprise Phone Configuration) or Common Phone Profile Configuration (Device > Device Settings > Common Phone Profile). (Switch Port Remote Configuration or PC Port Remote Configuration). |

| Feature | Description | Configuration reference |
|---|---|---|
| Shared Line | Allows multiple phones to share the same phone number or allows a user to share a phone number with a coworker. | For more information, see the <i>Cisco Unified Communications Manager System Guide</i> , “Understanding Directory Numbers” chapter. |
| Speed Dialing (Cisco Unified IP Phone 6911 only) | Dials a specified number that has been previously stored. You configure this feature on the Cisco Unified Communications Manager. The users access speed dialing using the Feature key and the Feature Code. | For more information see: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, “Cisco Unified IP Phone Configuration” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Cisco Unified IP Phone” chapter |
| SSH Access | Allows the administrator to enable or disable the SSH Access setting using Cisco Unified CM Administration. Enabling Secure Shell (SSH) access allows the phone to accept the SSH connections. Disabling SSH access blocks SSH access to the phone. | For more information, see UCR 2008 Setup, on page 68 |
| Time-of-Day Routing | Restricts access to specified telephony features by time period. | For more information, see: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, “Time Period Configuration” chapter • <i>Cisco Unified Communications Manager System Guide</i>, “Time-of-Day Routing” chapter |
| Time Zone Update | Updates the Cisco Unified IP Phone with time zone changes. | For more information, see the <i>Cisco Unified Communications Manager Administration Guide</i> , “Date/Time Group Configuration” chapter. |
| Transfer | Allows users to redirect connected calls from their phones to another number. | Some JTAPI/TAPI applications are not compatible with the Direct Transfer feature implementation on the Cisco Unified IP Phone. |

| Feature | Description | Configuration reference |
|--------------------------------|--|--|
| UCR 2008 (SCCP phones only) | <p>The Cisco Unified IP Phones using SCCP support Unified Capabilities Requirements (UCR) 2008 by providing the following functions:</p> <ul style="list-style-type: none"> • Support for Federal Information Processing Standard (FIPS) 140-2 • Support for 80-bit SRTCP Tagging <p>As an IP Phone administrator, you must set up specific parameters in Cisco Unified Communications Manager Administration.</p> | See UCR 2008 Setup , on page 68. |
| Voice messaging system | <p>Enables callers to leave messages if calls are unanswered.</p> <p>Users access messages using the Message button on the Cisco Unified IP Phone 6911.</p> <p>Users access messages using the access code on the Cisco Unified IP Phone 6901.</p> | <p>For more information, see:</p> <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, “Cisco Voice-Mail Port Configuration” chapter. • <i>Cisco Unified Communications Manager System Guide</i>, “Voice Mail Connectivity to Cisco Unified Communications Manager” chapter. |

Add Users to Cisco Unified Communications Manager

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

- Create a personal directory.
- Set up speed dial and call forwarding numbers.

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

Procedure

- To add users individually, choose **User Management > End User** from Cisco Unified Communications Manager.
For more information, see the “End User Configuration” chapter in the *Cisco Unified Communications Manager Administration Guide*.
- To add users in batches, use the Bulk Administration Tool. This method also enables you to set an identical default password for all users.
For more information, see the “Bulk Administration” chapter in *Cisco Unified Communications Manager Administration Guide*.
- To add users from your corporate Lightweight Directory Access Protocol (LDAP) directory, choose **System > LDAP > LDAP System** from Cisco Unified Communications Manager.



Note After you enable synchronization with the LDAP server, you cannot add additional users from Cisco Unified Communications Manager Administration.

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

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

User Options Web Pages Management

From the User Options web page, users can customize and control several phone features and settings. For more information about User Options web pages, see *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.

User Access to User Options Web Pages

Before a user can access the User Options web pages, you must add the user to the standard Cisco Unified Communications Manager End User group and associate the appropriate phone with the user.

Make sure to provide users with the following information about the User Options web pages:

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

http://<server_name:portnumber>/ccmuser/, where *server_name* is the host name of the Cisco Unified Communications Manager.

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

These settings correspond to the values you enter when you add the user to Cisco Unified Communications Manager (see the [Add Users to Cisco Unified Communications Manager, on page 62](#)).

For more information, see:

- *Cisco Unified Communications Manager Administration Guide*, “User Group Configuration” chapter.
- *Cisco Unified Communications Manager Administration Guide*, “End User Configuration” chapter.

Add User to 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 Groups**. 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**.
-

Customize User Options Web Pages Options

Most options on the User Options web pages appear by default. However, the following options must be set by the system administrator by using Enterprise Parameters Configuration settings in Cisco Unified Communications Manager:

- Show Ring Settings

- Show Call Forwarding
- Show Message Waiting Lamp
- Show Audible Message Waiting Indicator



Note The settings apply to all User Options web pages at your site.

To specify the options that appear on the User Options web pages, perform these steps:

Procedure

- Step 1** From Cisco Unified Communications Manager, choose **System > Enterprise Parameters**.
The Enterprise Parameters Configuration window appears.
- Step 2** In the CCMUser Parameters area, specify whether a parameter appears on the User Options web pages by choosing one of the following values from the Parameter Value drop-down list:
- **True:** Option displays on the User Options web pages (default except for Show Ring Settings, and Show Call Forwarding).
 - **False:** Option does not display on the User Options web pages.
 - **Show All Settings:** All call forward settings display on the User Options web pages (default).
 - **Hide All Settings:** No call forward settings display on the User Options web pages.
 - **Show Only Call Forward All:** Only call forward all calls displays on the User Options web pages.

EnergyWise Setup on Cisco Unified IP Phone 6901 and 6911

To reduce power consumption, you can configure the phone to sleep (power down) and wake (power up) if your system includes an EnergyWise controller (for example, a Cisco Switch with the EnergyWise feature enabled).

You configure settings in Cisco Unified Communications Manager 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 sends back 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, reducing the power consumption to a predetermined level. A phone that is not idle sets an idle timer, and goes to sleep after the timer expires.

At the scheduled wake time, the system restores power to the phone, waking it up.

The following table explains the Cisco Unified Communications Manager fields that control the EnergyWise settings. You configure these fields in Cisco Unified Communications Manager in the Product Specific configuration window, by choosing **Device > Phone**.

Table 11: EnergyWise Configuration Fields

| Field | Description |
|------------------------|--|
| Enable Power Save Plus | <p>Selects the schedule of days for which the phone powers down. Select multiple days by pressing and holding the Control key while clicking on the days for the schedule.</p> <p>By default, no days are selected.</p> <p>When Enable Power Save is checked, you receive a message to warn about emergency (e911) concerns.</p> <p>Caution While Power Save Plus Mode (the mode) is in effect, endpoints 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 are taking 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 will fully inform users of the effects of the mode on calls, calling and otherwise.</p> <p>Note To disable Power Save Plus, you must uncheck the Allow EnergyWise Overrides check box. Leaving the Allow EnergyWise Overrides checked with no days selected in the Enable Power Save Plus field does not disable Power Save Plus.</p> |
| Phone On Time | <p>Determines when the phone automatically turns on for the days selected in the Enable Power Save Plus field.</p> <p>Enter the time in this field in 24 hour format, where 00:00 is midnight.</p> <p>For example, to automatically power up the phone at 7:00 a.m. (0700), enter 7:00. To power up the phone at 2:00 p.m. (1400), enter 14:00.</p> <p>The default value is 00:00.</p> |

| Field | Description |
|------------------------|--|
| Phone Off Time | <p>The time of day that the phone powers down for the days 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.</p> <p>Enter the time in this field in 24 hour format, where 00:00 is midnight.</p> <p>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.</p> <p>The default value is 24:00.</p> |
| Phone Off Idle Timeout | <p>The length of time that the phone must be idle before the phone powers down.</p> <p>The range of the field is 20 to 1440 minutes.</p> <p>The default value is 60 minutes.</p> |
| Enable Audible Alert | <p>When enabled, instructs the phone to play an audible alert starting at 10 minutes before to the time specified in the Phone Off Time field.</p> <p>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's designated volume level. The audible alert schedule is:</p> <ul style="list-style-type: none"> • 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 <p>This check box applies only if the Enable Power Save Plus list box has one or more days selected.</p> |
| EnergyWise Domain | <p>The EnergyWise domain that the phone is in. The maximum length is 127.</p> |
| EnergyWise Secret | <p>The security secret password that is used to communicate with the endpoints in the EnergyWise domain. The maximum length is 127.</p> |

| Field | Description |
|----------------------------|--|
| Allow EnergyWise Overrides | <p>This check box determines whether you will allow the EnergyWise domain controller policy to send power-level updates to the phones. The following conditions apply:</p> <ul style="list-style-type: none"> The settings in Cisco Unified CM Administration take effect on schedule even if EnergyWise sends an override. <p>For example, assume 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.</p> <ul style="list-style-type: none"> If EnergyWise directs the phone to turn off at 20:00 (8:00 p.m.), that directive will remain 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. To change the power level on the phone again, EnergyWise must reissue a new power-level change command. <p>Note To disable Power Save Plus, you must uncheck the Allow EnergyWise Overrides check box. Leaving the Allow EnergyWise Overrides checked with no days selected in the Enable Power Save Plus field does not disable Power Save Plus.</p> |

UCR 2008 Setup

The parameters that support UCR 2008 reside in Cisco Unified Communications Manager Administration. The following table describes the parameters and indicates the procedure to change the setting.

Table 12: UCR 2008 Parameter Location

| Parameter | Administration Path | Procedure |
|-----------|---|---|
| FIPS Mode | Device > Device Settings > Common Phone Profile | Set Up UCR 2008 in Common Phone Profile, on page 69 |
| | System > Enterprise Phone Configuration | Set Up UCR 2008 in Enterprise Phone Configuration, on page 70 |

| Parameter | Administration Path | Procedure |
|---|---|---|
| SSH Access | Device > Phone | Set Up UCR 2008 in Phone, on page 69 |
| | Device > Device Settings > Common Phone Profile | Set Up UCR 2008 in Common Phone Profile, on page 69 |
| Web Access | Device > Phone | Set Up UCR 2008 in Phone, on page 69 |
| 80-bit SRTCP | Device > Device Settings > Common Phone Profile | Set Up UCR 2008 in Common Phone Profile, on page 69 |
| | System > Enterprise Phone Configuration | Set Up UCR 2008 in Enterprise Phone Configuration, on page 70 |
| IP Addressing Mode | Device > Device Settings > Common Device Configuration | Set Up UCR 2008 in Common Device Configuration, on page 70 |
| IP Addressing Mode Preference for Signaling | Device > Device Settings > Common Device Configuration | Set Up UCR 2008 in Common Device Configuration, on page 70 |

Set Up UCR 2008 in Phone

Use this procedure to set the following UCR 2008 parameters:

- SSH Access
- Web Access

Procedure

-
- Step 1** Choose **Device > Phone**.
- Step 2** Set the SSH Access parameter to **Disabled**.
- Step 3** Set the Web Access parameter to **Disabled**.
- Step 4** Select **Save**.
-

Set Up UCR 2008 in Common Phone Profile

Use this procedure to set the following UCR 2008 parameters:

- FIPS Mode
- SSH Access
- 80-bit SRTCP

Procedure

- Step 1** Choose **Device > Device Settings > Common Phone Profile**.
 - Step 2** Set the FIPS Mode parameter to **Enabled**.
 - Step 3** Set the SSH Access parameter to **Disabled**.
 - Step 4** Set the 80-bit SRTCP parameter to **Enabled**.
 - Step 5** Select **Save**.
-

Set Up UCR 2008 in Enterprise Phone Configuration

Use this procedure to set the following UCR 2008 parameters:

- FIPS Mode
- 80-bit SRTCP

Procedure

- Step 1** Choose **System > Enterprise Phone Configuration**.
 - Step 2** Set the FIPS Mode parameter to **Enabled**.
 - Step 3** Set the 80-bit SRTCP parameter to **Enabled**.
 - Step 4** Select **Save**.
-

Set Up UCR 2008 in Common Device Configuration

Use this procedure to set the following UCR 2008 parameters:

- IP Addressing Mode
- IP Addressing Mode Preference for Signaling

Procedure

- Step 1** Choose **Device > Device Settings > Common Device Configuration**.
 - Step 2** Set the IP Addressing Mode parameter.
 - Step 3** Set the IP Addressing Mode Preference for Signaling parameter.
 - Step 4** Select **Save**.
-



CHAPTER 6

Self Care Portal Management

- [Self Care Portal Overview, on page 71](#)
- [Set Up User Access to the Self Care Portal, on page 71](#)
- [Customize the Self Care Portal Display, on page 72](#)

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.

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**.
-



CHAPTER 7

Remote Monitoring

- [Remote Monitoring Overview](#), on page 73
- [Cisco Unified IP Phone Web Pages](#), on page 73
- [Access Web Page for Phone](#), on page 74
- [Disable and Enable Web Page Access](#), on page 75
- [Device Information Area](#), on page 75
- [Network Setup Area](#), on page 76
- [Network Statistics Area](#), on page 81
- [Device Logs Area](#), on page 83
- [Streaming Statistics Area](#), on page 89

Remote Monitoring Overview

Each Cisco Unified IP Phone has a web page from which you can view information about the phone, that includes:

- Device information
- Network setup information
- Network statistics
- Device logs
- Streaming statistics

This chapter 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.

For more information about troubleshooting the Cisco Unified IP Phone, see [Troubleshooting and Maintenance](#), on page 93.

Cisco Unified IP Phone Web Pages

The web page for a Cisco Unified IP Phone includes these topics:

- **Device Information:** Displays device settings and related information for the phone.

- **Network Setup:** Displays network configuration information and information about other phone settings.
- **Network Statistics:** Displays information about network traffic using hyperlinks to the following windows:
 - **Ethernet Information:** Displays information about Ethernet traffic.
 - **Network (Port):** Displays information about network traffic to and from the network port on the phone.
- **Device Logs:** Displays information about troubleshooting using hyperlinks to the following windows:
 - **Console Logs:** Displays hyperlinks to individual log files.
 - **Core Dumps:** Displays hyperlinks to individual core dump files.
 - **Status Messages:** Displays up to 30 of the most recent status messages that the phone generated since it last powered up.
- **Streaming Statistics:** Displays information about stream statistics using hyperlinks to the following window:
 - **Stream:** Displays a variety of streaming statistics.

Related Topics

- [Device Information Area](#), on page 75
- [Network Setup Area](#), on page 76
- [Network Statistics Area](#), on page 81
- [Device Logs Area](#), on page 83
- [Streaming Statistics Area](#), on page 89

Access Web Page for Phone

To access the web page for a Cisco Unified IP Phone, perform these steps.



Note

If you cannot access the web page, it may be disabled. For more information, see the [Disable and Enable Web Page Access](#), on page 75.

Procedure

Step 1

Obtain the IP address of the Cisco Unified IP Phone using one of these methods:

- Search for the phone in Cisco Unified Communications Manager by choosing **Device > Phone**. Phones registered 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 Cisco Unified IP Phone, press the *, #, and 0 buttons simultaneously, enter the password, and then follow the voice prompts to review the network setting.

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

http://IP_address

Disable and Enable Web Page Access

For security purposes, you may choose to prevent access to the phone web pages. If you prevent access, the web pages described in this chapter and the Cisco Unified CM User Options web pages cannot display.

To disable access to the web pages for a phone, perform these steps from Cisco Unified Communications Manager.

Procedure

- Step 1** Choose **Device > Phone**.
- Step 2** Specify the criteria to find the phone and click **Find**, or click **Find** to display a list of all phones.
- Step 3** Click the device name to open the Phone Configuration window for the device.
- Step 4** Scroll down to the Product Specific Configuration section.
- Step 5** To disable access, from the Web Access drop-down list, choose Disabled.
- Step 6** To enable access, from the Web Access drop-down list, choose Enabled.
- Step 7** Click **Save**.
- Step 8** Click **Apply Config**.

Device Information Area

The Device Information area on a phone web page displays device settings and related information. The following table describes these items.

To display the Device Information area, access the web page for the phone as described in the [Access Web Page for Phone, on page 74](#), and then click the **Device Information** hyperlink.

Table 13: Device Information Area Items

| Item | Description |
|-------------|---|
| 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 assigned to the phone |
| App Load ID | Identifier of the firmware running on the phone |

| Item | Description |
|-------------------|--|
| Boot Load ID | Identifier of the factory-installed load running on the phone |
| Hardware Revision | Revision value of the phone hardware |
| Serial Number | Unique serial number of the phone |
| Model Number | Model number of the phone |
| Message Waiting | Indicates if there is a voice message waiting on the primary line for this phone |
| UDI | Displays the following Cisco Unique Device Identifier (UDI) information about the phone: <ul style="list-style-type: none"> • 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 Identifier: Represents the hardware version of the phone • Serial Number: Displays the unique serial number of the phone |
| Time | Time obtained from the Date/Time Group in Cisco Unified Communications Manager to which the phone belongs |
| Time Zone | Time zone obtained from the Date/Time Group in Cisco Unified Communications Manager to which the phone belongs |
| Date | Date obtained from the Date/Time Group in Cisco Unified Communications Manager to which the phone belongs |

Network Setup Area

The Network Setup 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 and the Phone Information Menu on the Cisco Unified IP Phone. For more information, see [Cisco Unified IP Phone Settings, on page 47](#).

To display the Network Setup area, access the web page for the phone as described in the [Access Web Page for Phone, on page 74](#), and click the **Network Configuration** hyperlink.

Table 14: Network Configuration Area Items

| Item | Description |
|------------------------|---|
| DHCP Server | IP address of the Dynamic Host Configuration Protocol (DHCP) server that assigns the phone IP address. |
| 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. |
| IP Address | Internet Protocol (IP) address of the phone. |
| Subnet Mask | Subnet mask used by the phone. |
| TFTP Server 1 | Primary Trivial File Transfer Protocol (TFTP) server used by the phone. |
| TFTP Server 2 | Backup Trivial File Transfer Protocol (TFTP) server used by the phone. |
| Default Router 1 | Default router used by the phone. |
| DNS Server 1 through 5 | Primary DNS server (DNS Server 1) and optional backup DNS servers (DNS Server 2 - 5) used by the phone. |
| Operational VLAN ID | Auxiliary Virtual Local Area Network (VLAN) configured on a Cisco Catalyst switch in which the phone is a member. |
| Admin. VLAN ID | Auxiliary VLAN in which the phone is a member. |

| Item | Description |
|--|---|
| CallManager 1–5 | <p>Host names or IP addresses, in prioritized order, of the Cisco Unified Communications Manager servers with which the phone can register.</p> <p>For an available server, an item shows the Cisco Unified Communications Manager server IP address and one of the following states:</p> <ul style="list-style-type: none"> • Active: Cisco Unified Communications Manager server from which the phone is currently receiving call processing services. • Standby: Cisco Unified Communications Manager server to which the phone switches if the current server becomes unavailable. • Blank: No current connection to the Cisco Unified Communications Manager server. <p>An item may also include the Survivable Remote Site Telephony (SRST) designation, which identifies an SRST router capable of providing Cisco Unified Communications Manager functionality with a limited feature set. This router assumes control of call processing if all other Cisco Unified Communications Manager servers become unreachable. The SRST Cisco Unified Communications Manager always appears last in the list of servers, even if it is active. You configure the SRST router address in the Device Pool section in Cisco Unified Communications Manager Configuration window.</p> |
| DHCP Enabled | Indicates whether the phone uses DHCP. |
| DHCP Address Released | Indicates the setting of the DHCP Address Released option on the phone Network Configuration menu. |
| Alternate TFTP | Indicates whether the phone uses an alternative TFTP server. |
| Automatic Port Synchronization (Cisco Unified IP Phone 6911 only) | Indicates if the automatic port synchronization is enabled or disabled. When automatic port synchronization is enabled, Cisco recommends that you configure both ports to autonegotiate. If one port is enabled for autonegotiate and the other is at a fixed speed, the phone synchronizes to the fixed port speed. |
| SW Port Remote Configuration | Indicates if the software port configuration of the speed and duplex mode for the SW port is enabled or disabled. |

| Item | Description |
|--|--|
| PC Port Remote Configuration (Cisco Unified IP Phone 6911 only) | Indicates if the remote port configuration of the speed and duplex mode for the PC port is enabled or disabled. |
| SW Port Setup | Speed and duplex of the switch port, where: <ul style="list-style-type: none"> • A: Auto Negotiate • 10H: 10-BaseT/half duplex • 10F: 10-BaseT/full duplex • 100H: 100-BaseT/half duplex • 100F: 100-BaseT/full duplex • No Link: No connection to the switch port |
| PC Port Setup (Cisco Unified IP Phone 6911 only) | Speed and duplex of the switch port, where: <ul style="list-style-type: none"> • A: Auto Negotiate • 10H: 10-BaseT/half duplex • 10F: 10-BaseT/full duplex • 100H: 100-BaseT/half duplex • 100F: 100-BaseT/full duplex • No Link: No connection to the PC port |
| User Locale | User locale associated with the phone user. Identifies a set of detailed information to support users, including language, font, date and time formatting, and alphanumeric keyboard text information. |
| Network Locale | Network locale associated with the phone user. Identifies a set of detailed information to support the phone in a specific location, including definitions of the tones and cadences used by the phone. |
| User Locale Version | Version of the user locale loaded on the phone. |
| Network Locale Version | Version of the network locale loaded on the phone. |
| PC Port Disabled (Cisco Unified IP Phone 6911 only) | Indicates whether the PC port on the phone is enabled or disabled. |
| Speaker Enabled (Cisco Unified IP Phone 6911 only) | Indicates whether the speakerphone is enabled on the phone. |
| GARP Enabled | Indicates whether the phone learns MAC addresses from Gratuitous ARP (GARP) responses. |
| Video Capability Enabled (Cisco Unified IP Phone 6911 only) | Indicates whether the phone can participate in video calls when connected to an appropriately equipped PC. |

| Item | Description |
|---|---|
| Voice VLAN Enabled | Indicates whether the phone allows a device attached to the PC port to access the Voice VLAN. |
| DSCP for Call Control | Digital Signal Control Processing (DSCP) classification for call control signaling. |
| DSCP for Configuration | DSCP classification for any phone configuration transfer. |
| DSCP for Services | DSCP classification for phone-based services. |
| Security Mode | Displays the security mode for the phone. |
| Web Access Enabled | Indicates whether web access is enabled or disabled for the phone. |
| Span to PC Port (Cisco Unified IP Phone 6911 only) | Indicates whether the phone will forward packets transmitted and received on the network port to the access port. |
| CDP: PC Port (Cisco Unified IP Phone 6911 only) | <p>Indicates whether CDP is supported on the PC port (default is enabled).</p> <p>Enable CDP on the PC port when Cisco VT Advantage/Unified Video Advantage (CVTA) is connected to the PC port. CVTA does not work without CDP interaction with the phone.</p> <p>When CDP is disabled in Cisco Unified Communications Manager, a warning displays to indicate that disabling CDP on the PC port prevents CVTA from working.</p> <p>The current PC and switch port CDP values display on the Settings menu.</p> |
| CDP: SW Port | <p>Indicates whether CDP is supported on the switch port (default is enabled).</p> <p>Enable CDP on the switch port for VLAN assignment for the phone, power negotiation, QoS management, and 802.1x security.</p> <p>Enable CDP on the switch port when the phone is connected to a Cisco switch.</p> <p>When CDP is disabled in Cisco Unified Communications Manager, a warning displays to, indicate that CDP should be disabled on the switch port only if the phone is connected to a non-Cisco switch.</p> <p>The current PC and switch port CDP values are shown on the Settings menu.</p> |

| Item | Description |
|---|--|
| SSH Access Enabled | Indicates whether the phone accepts or blocks the SSH connections. |
| EnergyWise Level | Indicates the EnergyWise Level. |
| EnergyWise Domain | The EnergyWise domain that the phone is in. |
| Federal Information Processing Standard (FIPS) Mode Enabled | Indicates whether FIPS mode is enabled. |
| IP Addressing Mode | Displays the IP addressing mode that is available on the phone. |
| IP Preference Mode Control | Indicates the IP address version that the phone uses during signaling with Cisco Unified Communications Manager when both IPv4 and IPv6 are both available on the phone. |
| IPv6 Auto Configuration | Displays whether the autoconfiguration is enabled or disabled on the phone. |
| IPv6 CAPF Server | Common Name (from the Cisco Unified Communications Manager Certificate) of the CAPF used by the phone. |
| DHCPv6 | Dynamic Host Configuration Protocol version 6 (DHCPv6) automatically assigns IPv6 address to devices when you connect them to the network. Cisco Unified IP Phones enable DHCP by default. |
| IPv6 Default Router 1 | Default router used by the phone (Default Router 1). |
| IPv6 Address | The 128 bit IPv6 address of the phone. |
| IPv6 Prefix Length | Subnet prefix length that is used by the phone. The subnet prefix length is a decimal value from 1-128, that specifies the portion of the IPv6 address that comprises the subnet. |
| IPv6 TFTP Server | Indicates whether the phone is using the IPv6 Alternate TFTP server. |

Network Statistics Area

The Network Statistics area on a phone web page provides information about network traffic on the phone. To display a network statistics area, access the web page for the phone as described in the [Access Web Page for Phone, on page 74](#).

- Ethernet Information: Displays information about Ethernet traffic.

- Network Information (Cisco Unified IP Phone 6901): Displays information about network traffic to and from the network port (10/100 SW) on the phone.
- Network and Access Information (Cisco Unified IP Phone 6911): Displays information about network traffic to and from the network port (10/100 SW) on the phone.

The following sections describe the Ethernet Information and the Network Information areas.

Ethernet Information Area

The following table describes the fields in the Ethernet Information area.

Table 15: Ethernet Information Items

| Item | Description |
|---------------|---|
| Tx Frames | Total number of packets transmitted by the phone |
| Tx broadcast | Total number of broadcast packets transmitted by the phone |
| Tx multicast | Total number of multicast packets transmitted by the phone |
| Tx unicast | Total number of unicast packets transmitted by the phone |
| Rx Frames | Total number of packets received by the phone |
| Rx broadcast | Total number of broadcast packets received by the phone |
| Rx multicast | Total number of multicast packets received by the phone |
| Rx unicast | Total number of unicast packets received by the phone |
| RxPacketNoDes | Total number of shed packets caused by no Direct Memory Access (DMA) descriptor |

Network Information Area

The following table describes the fields in the Network Information area.

Table 16: Network Items on the Cisco Unified IP Phone 6901

| Item | Description |
|--------------|--|
| Tx Frames | Total number of packets transmitted by the phone |
| Tx broadcast | Total number of broadcast packets transmitted by the phone |

| Item | Description |
|---------------------------|---|
| Tx unicast | Total number of unicast packets transmitted by the phone |
| Rx Frames | Total number of packets received by the phone |
| Rx broadcast | Total number of broadcast packets received by the phone |
| Rx unicast | Total number of unicast packets received by the phone |
| Neighbor Device ID | Identifier of a device connected to this port discovered by CDP protocol or Link Layer Discovery Protocol (LLDP) |
| Neighbor IP Address | IP address of the neighbor device discovered by CDP protocol |
| Neighbor Port | Neighbor device port to which the phone is connected discovered by CDP protocol |
| LLDP AgeoutsTotal | Total number of LLDP frames that have been time out in cache |
| LLDP FramesDiscardedTotal | Total number of LLDP frames that are discarded when any of the mandatory TLVs is missing or out of order or contains out of range string length |
| LLDP FramesInErrorsTotal | Total number of LLDP frames that received with one or more detectable errors |
| LLDP FramesInTotal | Total number of LLDP frames received on the phone |
| LLDP TLVDiscardedTotal | Total number of LLDP Threshold Limit Values (TLV) that are discarded |
| LLDP TLVUnrecognizedTotal | Total number of LLDP TLVs that are not recognized on the phone |
| Restart Cause | Reason for the last restart |
| Port | Speed and duplex information |
| IPv4 | IPv4 Address |

Device Logs Area

The Device Logs area on a phone web page provides information you can use to monitor and troubleshoot the phone. To access a Device Log area, access the web page for the phone as described in the [Access Web Page for Phone, on page 74](#).

- **Console Logs:** Displays hyperlinks to individual log files. The console log files include debug and error messages received on the phone.
- **Core Dumps:** Displays hyperlinks to individual dump files. The core dump files include data from a phone crash.
- **Status Messages:** Displays up to the 80 most recent status messages that the phone has generated since it was last powered up. You can also see this information from the Status Messages screen on the Web page of the phone. [Status Messages Area, on page 84](#) describes the status messages that may be displayed.

The following sections describe the Status Messages Area.

Status Messages Area

The Status Messages web page displays up to 80 of the most recent status messages that the phone has generated since it was last powered up. You can access the Status Messages web page even if the phone is not running. The following table describes the status messages. This table also includes possible explanations and actions to troubleshoot errors.

Table 17: Status Messages on the Cisco Unified IP Phone 6900 Series

| Message | Description | Possible Explanation and Action |
|---------------------|---|--|
| CFG file not found | The name-based and default configuration file was not found on the TFTP Server. | <p>The Cisco Unified Communications Manager creates the configuration file for the phone when the phone is added to the database. If the phone has not been added to the Cisco Unified Communications Manager database, the TFTP server generates a CFG File Not Found response.</p> <p>Phone is not registered with Cisco Unified Communications Manager. You must manually add the phone to Cisco Unified Communications Manager if you are not allowing phones to autoregister. For more information, see the Cisco Unified Communications Manager Phone Addition Methods, on page 35.</p> <p>If you are using DHCP, verify that the DHCP server is pointing to the correct TFTP server.</p> <p>If you are using static IP addresses, check the TFTP server configuration.</p> |
| CFG TFTP Size Error | The configuration file is too large for the file system on the phone. | Power cycle the phone. |

| Message | Description | Possible Explanation and Action |
|------------------|--|--|
| Checksum Error | Downloaded software file is corrupted. | Obtain a new copy of the phone firmware and place it in the TFTPPath directory. You should only copy files into this directory when the TFTP server software is shut down, otherwise the files may be corrupted. |
| DHCP timeout | DHCP server did not respond. | <ul style="list-style-type: none"> • Network is busy: The errors should resolve themselves when the network load reduces. • No network connectivity between the DHCP server and the phone: Verify the network connections. • DHCP server is down: Check the DHCP server configuration. • Errors persist: Consider assigning a static IP address. |
| DNS timeout | DNS server did not respond. | <ul style="list-style-type: none"> • Network is busy: The errors should resolve themselves when the network load reduces. • No network connectivity between the DNS server and the phone: Verify the network connections. • DNS server is down: Check the DNS server configuration. |
| DNS unknown host | DNS could not resolve the name of the TFTP server or Cisco Unified Communications Manager. | <ul style="list-style-type: none"> • Verify that the host names of the TFTP server or Cisco Unified Communications Manager are configured properly in DNS. • Consider using IP addresses rather than host names. |
| Duplicate IP | Another device is using the IP address assigned to the phone. | <ul style="list-style-type: none"> • If the phone has a static IP address, verify that you have not assigned a duplicate IP address. • If you are using DHCP, check the DHCP server configuration. |

| Message | Description | Possible Explanation and Action |
|---------------------|--|--|
| 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 System Administration, check that the following files are located within subdirectories in the TFTP File Management: <ul style="list-style-type: none"> • Located in subdirectory with the same name as network locale: <ul style="list-style-type: none"> • tones.xml • Located in subdirectory with the same name as user locale: <ul style="list-style-type: none"> • glyphs.xml • dictionary.xml • kate.xml |
| File not found | The phone cannot locate the phone load file specified in the phone configuration file on the TFTP server. | From Cisco Unified Operating System Administration, make sure that the phone load file is on the TFTP server, and that the entry in the configuration file is correct. |
| IP address released | The phone has been configured to release the IP address. | The phone remains idle until it is power cycled or you reset the DHCP address. |
| Load ID incorrect | Load ID of the software file is of the wrong type. | Check the load ID assigned to the phone (from Cisco Unified Communications Manager, choose Device > Phone). Verify that the load ID is entered correctly. |
| Load rejected HC | The application that was downloaded is not compatible with the phone hardware. | Occurs if you were attempting to install a version of software on this phone that did not support hardware changes on this newer phone. Check the load ID assigned to the phone (from Cisco Unified Communications Manager, choose Device > Phone). Re-enter the load displayed on the phone. |

| Message | Description | Possible Explanation and Action |
|---------------------|--|--|
| No default router | The DHCP or static configuration did not specify a default router. | <ul style="list-style-type: none"> • If the phone has a static IP address, verify that the default router has been configured. • If you are using DHCP, the DHCP server may not contain default router. Check the DHCP server configuration. |
| No DNS server IP | A name was specified but the DHCP or static IP configuration did not specify a DNS server address. | <ul style="list-style-type: none"> • If the phone has a static IP address, verify that the DNS server has been configured. • If you are using DHCP, the DHCP server may not contain a DNS server. Check the DHCP server configuration. |
| TFTP access error | TFTP server is pointing to a directory that does not exist. | <ul style="list-style-type: none"> • If you are using DHCP, verify that the DHCP server is pointing to the correct TFTP server. • If you are using static IP addresses, check the TFTP server configuration. |
| TFTP error | The phone does not recognize an error code provided by the TFTP server. | Contact the Cisco TAC. |
| TFTP file not found | The requested load file (.bin) was not found in the TFTPPath directory. | Check the load ID assigned to the phone. From Cisco Unified Communications Manager, choose Device > Phone . Verify that the TFTPPath directory contains a .bin file with this load ID as the name. |

| Message | Description | Possible Explanation and Action |
|--|---|--|
| TFTP server not authorized | The specified TFTP server could not be found in the phone CTL. | <ul style="list-style-type: none"> The DHCP server has the wrong configuration file for the TFTP server. Update the TFTP server configuration to specify the correct TFTP server. The CTL file was made and then the TFTP server address changed. Regenerate the CTL file. If the phone is using a static IP address, the phone may be configured with the wrong TFTP server address. Enter the correct TFTP server address in the Network Configuration menu on the phone. If the TFTP server address is correct, there may be a problem with the CTL file. Run the CTL client and update the CTL file, making sure that the proper TFTP servers are included in this file. |
| TFTP timeout | TFTP server did not respond. | <ul style="list-style-type: none"> Network is busy: The errors should resolve themselves when the network load reduces. No network connectivity between the TFTP server and the phone: Verify the network connections. TFTP server is down: Check the TFTP server configuration. |
| Timed Out | Supplicant attempted 802.1X transaction but timed out to due the absence of an authenticator. | Authentication typically times out if 802.1X is not configured on the switch. |
| Version error | The name of the phone load file is incorrect. | Make sure that the phone load file has the correct name. |
| XmlDefault.cnf.xml, or .cnf.xml corresponding to the phone device name | Name of the configuration file. | None. This is an informational message indicating the name of the configuration file for the phone. |

Streaming Statistics Area

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 running a service that sends or receives audio or data.

The Streaming Statistics areas on a phone web page provide information about the streams. Cisco Unified IP Phone 6900 Series phones use only Stream 1.

To display a Streaming Statistics area, access the web page for the phone as described in the [Access Web Page for Phone, on page 74](#), and then click the **Stream 1** hyperlink.

The following table describes the items in the Streaming Statistics areas.

Table 18: Streaming Statistics Area Items

| Item | Description |
|-------------------------|--|
| Remote Address | IP address and UDP port of the destination of the stream. |
| Local Address | IP address and UDP port of the phone. |
| Start Time | Internal time stamp indicating when Cisco Unified Communications Manager requested that the 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 address. |
| Sender Packets | Total number of RTP data packets transmitted by the phone since starting this connection. The value is 0 if the connection is in receive-only mode. |
| Sender Octets | Total number of payload octets transmitted in RTP data packets by the phone since starting this connection. Receive-only connections display 0 in the field. |
| Sender Codec | Type of audio encoding used for the transmitted stream. |
| Sender Reports Sent | Number of times the RTCP Sender Report have been sent. |
| Sender Report Time Sent | Internal time stamp of when the last RTCP Sender Report was sent. |

| Item | Description |
|-----------------------------|--|
| Rcvr Lost Packets | Total number of RTP data packets that have been lost since starting receiving data on this connection. Defined as the number of expected packets less the number of packets actually received, where the number of received packets includes any that are late or duplicate. Send-only connections display 0 in the field. |
| Avg Jitter | Estimate of mean deviation of the RTP data packet interarrival time, measured in milliseconds. Send-only connections display 0 in the field. |
| Rcvr Codec | Type of audio encoding used for the received stream. |
| Rcvr Reports Sent | Number of times the RTCP Receiver Reports have been sent. |
| Rcvr Report Time Sent | Internal time stamp indication when a RTCP Receiver Report was sent. |
| Rcvr Packets | Total number of RTP data packets received by the phone since starting receiving data on this connection. Includes packets received from different sources if this is a multicast call. Send-only connections display 0 in the field. |
| Rcvr Octets | Total number of payload octets received in RTP data packets by the device since starting reception on the connection. Includes packets received from different sources if this is a multicast call. Send-only connections display 0 in the field. |
| 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. |
| Max Jitter | Maximum value of instantaneous jitter, in milliseconds. |
| Sender Size | RTP packet size, in milliseconds, for the transmitted stream. |
| Sender Reports Received | Number of times RTCP Sender Reports have been received. |
| Sender Report Time Received | Last time at which an RTCP Sender Report was received. |
| Rcvr Size | RTP packet size, in milliseconds, for the received stream. |

| Item | Description |
|---------------------------|--|
| Rcvr Discarded | RTP packets received from network but discarded from jitter buffers. |
| Rcvr Reports Received | Number of times RTCP Receiver Reports have been received. |
| Rcvr Report Time Received | Last time at which an RTCP Receiver Report was received. |



Note When the RTP Control Protocol is disabled, no data generates for this field and thus displays as 0.

Related Topics

[Cisco Unified IP Phone Settings](#), on page 47



CHAPTER 8

Troubleshooting and Maintenance

- [Troubleshooting and Maintenance Overview, on page 93](#)
- [Troubleshooting, on page 93](#)
- [Maintenance, on page 105](#)

Troubleshooting and Maintenance Overview

This chapter provides information that can assist you in troubleshooting problems with your Cisco Unified IP Phone or with your IP telephony network. It also explains how to clean and maintain your phone.

If you need additional assistance to resolve an issue, see the [Documentation, Support, and Security Guidelines, on page xiii](#).

Troubleshooting

Use the following sections to troubleshoot problems with the phones.

Startup Problems

After installing a Cisco Unified IP Phone into your network and adding it to Cisco Unified Communications Manager, the phone should start up as described in the [Phone Startup Verification, on page 45](#).

If the phone does not start up properly, see the following sections for troubleshooting information.

Cisco Unified IP Phone Does Not Go Through Normal Startup Process

Problem

When you connect a Cisco Unified IP Phone into the network port, the phone should go through the normal startup process as described in [Phone Startup Verification, on page 45](#).

Cause

If the phone does not go through the startup process, the cause may include faulty cables, bad connections, network outages, and lack of power. Or, the phone may be faulty.

Solution

To determine whether the phone is faulty, follow these suggestions to systematically eliminate other potential problems:

- Verify that the network port is functional:
 - Exchange the Ethernet cables with cables that you know are functional.
 - Connect a operational phone to this network port to verify that the port is active.
 - Replace an operational phone with the nonoperational phone.
 - Connect the nonoperational phone 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 has power.
 - If you are using in-line power, plug the phone into an electrical outlet using the external power supply.
 - If you are using the external power supply, switch the power supply with a unit that you know works.
- If the phone still does not start up properly, perform a factory reset of the phone. For instructions, see the [Perform Factory Reset, on page 106](#).

If after attempting these solutions, the phone still does not function, contact a Cisco technical support representative for additional assistance.

Cisco Unified IP Phone Does Not Register with Cisco Unified Communications Manager

Use this section if the phone proceeds past the first stage of the startup process (all LED buttons on) but the phone is not starting up properly. The phone cannot successfully start up unless it is connected to the Ethernet network and it has registered with a Cisco Unified Communications Manager server.

In addition, problems with security may prevent the phone from starting up properly. For more information, see [General Troubleshooting Information, on page 103](#).

Phone Displays Error Messages**Problem**

Status messages display errors during startup.

Solution

As the phone cycles through the startup process, you can access status messages that might provide you with information about the cause of a problem. For instructions about accessing status messages and suggested actions to resolve the errors, see the [Device Logs Area, on page 83](#).

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.

TFTP Server Settings

Problem

The TFTP server settings may not be correct.

Solution

Check the TFTP settings. See [Check TFTP Settings, on page 101](#).

IP Addressing and Routing

Problem

The IP addressing and routing fields may not be correctly configured.

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.

On the Cisco Unified IP Phone, press the *, #, and 0 buttons simultaneously, enter the password, and then follow the voice prompts to review the 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. See the [Cisco Unified IP Phone Network Settings Setup, on page 47](#) for instructions.
- If you are using DHCP, check the IP addresses distributed by your DHCP server. Refer to 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

Cisco CallManager and TFTP Services Are Not Running

Problem

If the Cisco CallManager 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 CallManager 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 103](#).

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. See [Create New Configuration File, on page 101](#).

Cisco Unified Communications Manager Phone Registration**Problem**

The phone is not registered with the Cisco Unified Communications Manager.

Solution

A Cisco Unified IP Phone can register with a Cisco Unified Communications Manager server only if the phone has been added to the server or (if autoregistration is enabled) there is a sufficient number of unit licenses. Review the information and procedures in [Cisco Unified Communications Manager Phone Addition Methods, on page 35](#) to ensure that the phone has been added to the Cisco Unified Communications Manager database.

To verify that the phone is in the Cisco Unified Communications Manager database, choose **Device > Phone > Find** from Cisco Unified Communications Manager Administration to search for the phone based on the MAC Address. For information about determining a MAC address, see the [Cisco Unified IP Phones and Different Protocols, on page 38](#).

If the phone is already in the Cisco Unified Communications Manager database, the phone configuration file may be damaged. For assistance, see [Create New Configuration File, on page 101](#).

For more information about licensing, see the “Licences for phones” section of the *Cisco Unified Communications Manager System Guide*.

Cisco Unified 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.

Cisco Unified IP Phone Displays Flashing Red Light

Problem

The phone fails to boot, and the message indicator flashes a red light.

Cause

When a Cisco Unified IP Phone boots, it performs an internal Power On Self Test (POST). POST checks for existing encryption functionality. If POST detects that encryption functionality is missing, the phone fails to boot, and the phone displays a flashing red light.

Solution

To correct the problem, perform the following steps:

1. Reset the phone manually.
2. If the phone does not start up properly, power up the phone with the handset off-hook. When the phone is powered up in this way, it attempts to launch a backup software image.
3. If the phone still does not start up properly, perform a factory reset of the phone. For instructions, see [Perform Factory Reset, on page 106](#).

Cisco Unified IP Phone Resets Unexpectedly

If users report that their phones are resetting during calls or while idle on their desk, you should investigate the cause. If the network connection and Cisco Unified Communications Manager connection are stable, a Cisco Unified IP Phone should not reset on its own.

Typically, a phone resets if it has problems connecting to the Ethernet network or to Cisco Unified Communications Manager.

Physical Connection Problems

Problem

The physical connection to the LAN may be broken.

Solution

Verify that the Ethernet connection to which the Cisco Unified 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.

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.

DHCP Settings Errors

Problem

The DHCP settings may be incorrect.

Solution

Verify that you have properly configured the phone to use DHCP. For more information see, [Cisco Unified IP Phone Network Settings Setup, on page 47](#). Verify that the DHCP server is set up properly. Verify the DHCP lease duration. Cisco recommends that you set the lease duration to 8 days.

Static IP Address Settings Errors

Problem

The static IP address assigned to the phone may be incorrect.

Solution

If the phone has been assigned a static IP address, verify that you have entered the correct settings. For more information, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#).

Voice VLAN Setup Errors

Problem

If the Cisco Unified IP Phone appears to reset during heavy network usage (for example, following extensive web surfing on a computer connected to the same switch as the phone), 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. For more information, see [Cisco Unified IP Phone 6911 and VLAN Interaction, on page 30](#).

Phones Have Not Been Intentionally 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 Unified IP Phone received a command from Cisco Unified Communications Manager to reset by pressing **Applications** on the phone and choosing **Administrator Settings > Status > Network Statistics**.

- If the Restart Cause field displays `Reset-Reset`, the phone received a Reset/Reset from Cisco Unified Communications Manager Administration.
- If the Restart Cause field displays `Reset-Restart`, the phone reset because it received a Reset/Restart from Cisco Unified Communications Manager Administration.

DNS or Other Connectivity Errors

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 with [Determine DNS or Connectivity Issues, on page 102](#).

Power Connection Problems

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.

Audio Problems

The following sections describe how to resolve audio problems.

Poor Audio Quality with Calls That Route Outside Cisco Unified Communications Manager

Problem

Poor quality occurs with tandem audio encoding. Tandem encoding can occur when calls are made between an IP Phone and a digital cellular phone, when a conference bridge is used, or in situations where IP-to-IP calls are partially routed across the PSTN.

Cause

In these cases, use of voice codecs such as G.729 and iLBC may result in poor voice quality.

Solution

Use the G.729 and iLBC codecs only when absolutely necessary.

Choppy Speech

Problem

A user complains of choppy speech on a call.

Cause

There may be a mismatch in the jitter configuration.

Solution

Check the AvgJtr and the MaxJtr statistics. A large variance between these statistics might indicate a problem with jitter on the network or periodic high rates of network activity.

No Speech Path

Problem

One or more people on a call do not hear any audio.

Solution

When at least one person in a call does not receive audio, IP connectivity between phones is not established. Check the configuration of routers and switches to ensure that IP connectivity is properly configured.

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.

Check TFTP Settings

Procedure

-
- Step 1** Determine the IP address of the TFTP server used by the phone by pressing the *, #, and 0 buttons simultaneously, entering the password, and then following the voice prompts to review the network setting.
 - Step 2** If you have assigned a static IP address to the phone, check the setting for the TFTP Server 1 option. See the [Cisco Unified IP Phone Network Settings Setup, on page 47](#).
 - Step 3** If you are using DHCP, the phone obtains the address for the TFTP server from the DHCP server. Check the IP address configured in Option 150 or Option 66.
 - Step 4** Enable the phone to use an alternate TFTP server. Such a setting is particularly useful if the phone was recently moved from one location to another. See the [Cisco Unified IP Phone Network Settings Setup, on page 47](#) for instructions.
-

Create New Configuration File

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.

**Note**

- When you remove a phone from the Cisco Unified Communications Manager database, the Cisco Unified Communications Manager deletes the configuration file from the TFTP server. The assigned directory numbers for the phone remain in the Cisco Unified Communications Manager database. They are called *unassigned DNs* and can be assigned to other devices. If unassigned DNs are not used by other devices, delete them from the Cisco Unified Communications Manager database. You can use the Route Plan Report to view and delete unassigned reference numbers. Refer to For more information, see *Cisco Unified Communications Manager Administration Guide*.
- 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 there is no button on the phone for the line, meaning that no calls to the number can be answered. These directory numbers should be removed from the phone and deleted if necessary.

To create a new configuration file, follow these steps:

Procedure

-
- Step 1** From Cisco Unified Communications Manager, choose **Device > Phone > Find** to locate the phone experiencing problems.
 - Step 2** Choose **Delete** to remove the phone from the Cisco Unified Communications Manager database.
 - Step 3** Add the phone back to the Cisco Unified Communications Manager database. See the [Cisco Unified Communications Manager Phone Addition Methods, on page 35](#) for details.
 - Step 4** Power cycle the phone.
-

Determine DNS or Connectivity Issues

If the phone continues to reset, follow these steps to eliminate DNS or other connectivity errors:

Procedure

-
- Step 1** Use the Reset Settings menu to reset phone settings to their default values. For more information, see [Cisco Unified IP Phone Reset or Restore, on page 105](#).
 - Step 2** Modify DHCP and IP settings:
 - a) Disable DHCP. For instructions, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#).
 - b) Assign static IP values to the phone. For instructions, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#). Use the same default router setting used for other functioning Cisco Unified IP Phones.
 - c) Assign a TFTP server. For instructions, see [Cisco Unified IP Phone Network Settings Setup, on page 47](#). Use the same TFTP server used for other functioning Cisco Unified IP Phones.
 - 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 the server is referred to by the IP address and not by the DNS name.

- Step 5** From Cisco Unified Communications Manager, choose **Device > Phone > Find** and verify that you have assigned the correct MAC address to this Cisco Unified IP Phone. For information about determining a MAC address, see the [Cisco Unified IP Phones and Different Protocols, on page 38](#).
- Step 6** Power cycle the phone.

Start Service



Note A service must be activated before it can be started or stopped. To activate a service, choose **Tools > Service Activation**.

To start a service, follow these steps:

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.

General Troubleshooting Information

The following table provides general troubleshooting information for the Cisco Unified IP Phone.

Table 19: Cisco Unified IP Phone Troubleshooting

| Summary | Explanation |
|---|--|
| Connecting a Cisco Unified IP Phone to another Cisco Unified IP Phone through the PC port. | Cisco does not support connecting an IP Phone to another IP Phone through the PC port. Each IP Phone should directly connect to a switch port. If phones are connected together in a line (by using the PC port), the phones will 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 voice VLAN may cause IP phones to reset, lose an active call, or be unable to initiate or answer a call. Phones may not regain connectivity until a broadcast storm ends. |

| Summary | Explanation |
|---|---|
| <p>Moving a network connection from the phone to a workstation.</p> | <p>If you power your phone through the network connection, you must be careful if you decide to unplug the network connection of the phone and plug the cable into a desktop computer.</p> <p>Caution The computer network card cannot receive power through the network connection; if power comes through the connection, the network card can be destroyed. To protect a network card, wait 10 seconds or longer after unplugging the cable from the phone before plugging it into a computer. This delay gives the switch enough time to recognize that there is no longer a phone on the line and to stop providing power to the cable.</p> |
| <p>Changing the telephone configuration.</p> | <p>By default, the network configuration options are locked to prevent users from making changes that could impact their network connectivity. You must unlock the network configuration options before you can configure them. See the Access Phone Configuration Settings, on page 48 for details.</p> |
| <p>Phone resetting.</p> | <p>The phone resets when it loses contact with the Cisco Unified Communications Manager software. This lost connection can be due to any network connectivity disruption, including cable breaks, switch outages, and switch reboots.</p> |
| <p>Codec mismatch between the phone and another device.</p> | <p>The RxType and the TxType statistics show the codec that is being used for a conversation between this Cisco Unified IP phone and the other device. The values of these statistics should match. If they do not match, verify that the other device can handle the codec conversation or that a transcoder is in place to handle the service.</p> |
| <p>Sound sample mismatch between the phone and another device.</p> | <p>The RxSize and the TxSize statistics show the size of the voice packets that are being used in a conversation between this Cisco Unified IP phone and the other device. The values of these statistics should match.</p> |

| Summary | Explanation |
|----------------------------|---|
| <p>Loopback condition.</p> | <p>A loopback condition can occur when the following conditions are met:</p> <ul style="list-style-type: none"> • The SW Port Configuration option in the Network Configuration menu on the 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) <p>In this case, the switch port on the phone can become disabled and the following message will appear in the switch console log:</p> <pre>HALF_DUX_COLLISION_EXCEED_THRESHOLD</pre> <p>To resolve this problem, reenable the port from the switch.</p> |

Additional Troubleshooting Information

If you have additional questions about troubleshooting the Cisco Unified IP Phones, several Cisco.com web sites can provide you with more tips. Choose from the sites available for your access level.

- Cisco Unified IP Phone Troubleshooting Resources:
http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_troubleshoot_and_alerts.html
- Cisco Products and Services (Technical Support and Documentation):
http://www.cisco.com/en/US/products/ps10326/tsd_products_support_series_home.html

Maintenance

The following sections describe voice and phone maintenance.

Cisco Unified IP Phone Reset or Restore

The following sections detail two general methods for resetting or restoring the Cisco Unified IP Phone.

Perform Basic Reset

Performing a basic reset of a Cisco Unified IP Phone provides a methods to recover if the phone experiences an error and provides a methods to reset or restore various configuration and security settings.

The following table describes the methods 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 20: Basic Reset Methods

| Operation | Performing | Explanation |
|----------------|---|--|
| Restart phone | Unplug the power cable and plug it back in. | 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, then restarts the phone. |
| Reset Settings | Reset the phone to the factory settings. | Resets user and network configuration settings to their default values, and restarts the phone. |

Perform Factory Reset

When you perform a factory reset of the Cisco Unified IP Phone, the following information is erased or reset to the default value:

- User configuration settings: Reset to default values
- Network configuration settings: Reset to default values
- Call histories: Erased
- Locale information: Reset to default values
- Phone application: Erased (phone recovers by using the image in the inactive partition of flash memory to boot up).

Before you perform a factory reset, ensure that the following conditions are met:

- The phone must be on a DHCP-enabled network.
- A valid TFTP server must be set in DHCP option 150 or option 66 on the DHCP server.

To perform a factory reset of a phone, you can use the IVR to reset the network settings to factory default or you can perform the following steps:

Procedure

-
- Step 1** Unplug the power cable from the phone and then plug it back in.
The phone begins its power-up cycle.
- Step 2** While the phone is powering up, press and hold # button until the Line LED turns green.
- Step 3** Release the # button and press **123456789*0#**.
The line button LED turns red. The phone reboots when it is finished.
-

Voice Quality Monitoring

To measure the voice quality of calls that are sent and received within the network, Cisco Unified IP Phones use statistical metrics that are based on concealment events. The Digital Signal Processor (DSP) plays concealment frames to mask frame loss in the voice packet stream.

- **Concealment Ratio metrics:** Shows the ratio of concealment frames over total speech frames. An interval conceal ratio is calculated every 3 seconds.
- **Concealed Second metrics:** Shows 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.
- **Mean Opinion Score (MOS) for Listening Quality (LQK) Voice Metrics:** Uses a numeric score to estimate the relative voice-listening quality. The Cisco Unified IP Phones calculate the MOS LQK based on audible-concealment events due to a frame loss in the preceding 8 seconds and includes weighting factors such as codec type and frame size.

MOS LQK scores are produced by a Cisco-proprietary algorithm, the Cisco Voice Transmission Quality (CVTQ) index. Depending on the MOS LQK version number, these scores may comply with the International Telecommunications Union (ITU) standard P.564. This standard defines evaluation methods and performance accuracy targets that predict listening quality scores based on observation of actual network impairment.



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 remotely by using Streaming Statistics (see [Remote Monitoring, on page 73](#)).

Voice Quality Metrics

When using the metrics for monitoring voice quality, note the typical scores under normal conditions of zero packet loss and use the metrics as a baseline for comparison.

It is also important to distinguish significant changes from random changes in metrics. Significant changes are scores that change about 0.2 MOS or more and persist in calls that last longer than 30 seconds. Conceal ratio changes indicate a frame loss greater than 3 percent.

The MOS LQK scores can vary based on the codec that the Cisco Unified IP Phone uses. The following codecs provide these corresponding maximum MOS LQK scores under normal conditions with zero frame loss for Cisco Unified Phones 6901 and 6911:

- G.711: 4.5 MOS LQK
- G.722: 4.5 MOS LQK
- G.728/iLBC: 3.9 MOS LQK
- G729A/AB: 3.7 MOS LQK

Cisco Voice Transmission Quality (CVTQ) does not support wideband (7 kHz) speech codecs, because ITU has not defined the extension of the technique to wideband. Therefore, MOS LQK scores that correspond to

G.711 performance are reported for G.722 calls to allow basic quality monitoring, rather than not reporting an MOS score.

- Reporting G.711-scale MOS scores for wideband calls through the use of CVTQ allows basic-quality classifications to be indicated as good/normal or bad/abnormal. Calls with high scores (approximately 4.5) indicate high quality or a low packet loss, and lower scores (approximately 3.5) indicate low quality or a high packet loss.
- Unlike MOS, the conceal ratio and concealed seconds metrics remain valid and useful for both wideband and narrowband calls.

A conceal ratio of zero indicates that the IP network is delivering frames and packets on time with no loss.

Voice Quality Troubleshooting Tips

When you observe significant and persistent changes to metrics, use the following table for general troubleshooting information.

Table 21: 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. | <ul style="list-style-type: none"> • 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. <p>Check packet transmit (TxCnt) and packet receive (RxCnt) counters to verify that voice packets are flowing.</p> |
| MOS LQK scores decrease significantly | <p>Network impairment from packet loss or high jitter levels:</p> <ul style="list-style-type: none"> • Average MOS LQK decreases may indicate widespread and uniform impairment. • Individual MOS LQK decreases may indicate bursty impairment. <p>Cross-check the conceal ratio and conceal seconds for evidence of packet loss and jitter.</p> |
| MOS LQK scores increase significantly | <ul style="list-style-type: none"> • 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 Unified IP Phone Cleaning

To clean your Cisco Unified IP Phone, use only a dry soft cloth to gently wipe the phone. Do not apply liquids or powders directly on the phone. As with all non-weatherproof electronics, liquids and powders can damage the components and cause failures.



APPENDIX A

Internal Support Web Site

- [Internal Web Site Overview](#), on page 111
- [Cisco Unified IP Phone User Support](#), on page 111
- [User Options Web Pages Access](#), on page 111
- [Phone Features User Subscription and Setup](#), on page 112
- [User Voice Messaging System Access](#), on page 112

Internal Web Site Overview

If you are a system administrator, you are likely the primary source of information for Cisco Unified IP Phone users in your network or company. It is important to provide current and thorough information to users.

Cisco recommends that you create a web page on your internal support site that provides users with important information about their Cisco Unified IP Phones.

Cisco Unified IP Phone User Support

To successfully use some of the features on the Cisco Unified IP Phone (including speed dial and voice message system options), users must receive information from you or from your network team or 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.

User Options Web Pages Access

Before a user can access the User Options web pages, you must use Cisco Unified Communications Manager Administration to add the user to a standard Cisco Unified Communications Manager End User group: choose **User Management > User Groups**. For more information, see:

- *Cisco Unified Communications Manager Administration Guide*, “User Group Configuration” chapter
- *Cisco Unified Communications Manager System Guide*, “Roles and User Groups” chapter

Phone Features User Subscription and Setup

Users can perform a variety of activities using the Cisco Unified Communications Manager User Options web pages. These activities include setting up speed dial on the Cisco Unified IP Phone 6911, setting up call forwarding numbers on the Cisco Unified IP Phone 6901 and 6911, and creating a personal address book. Keep in mind that configuring settings on a phone using a website might be new for your users. You need to provide as much information as possible to ensure that they can successfully access and use the User Options web pages.

Make sure to provide users with the following information about the User Options web pages:

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

http://<server_name:portnumber>/ccmuser/, where *server_name* is the host on which the web server is installed.

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

These settings correspond to the values you entered when you added the user to Cisco Unified Communications Manager (see the [Add Users to Cisco Unified Communications Manager](#), on page 62).

- A brief description of what a web-based, graphical user interface application is, and how to access it with a web browser.
- An overview of the tasks that users can accomplish using the web page.

For information using the User Options web pages, see *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.

User Voice Messaging System Access

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

You should provide this information to each user:

- How to access the voice messaging system account.

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

- Initial password for accessing the voice messaging system.

Make sure that you have configured a default voice messaging system password for all users.

- How the phone indicates that voice messages are waiting.

Make sure that you have used Cisco Unified Communications Manager to set up a message waiting indicator (MWI) method.



APPENDIX **B**

International User Support

- [International User Support Overview](#), on page 113
- [Unified Communications Manager Endpoints Locale Installer](#), on page 113

International User Support Overview

Localized versions of the Cisco Unified IP Phones are available. If you are supporting Cisco Unified IP Phones in a non-English environment, refer to the following sections to ensure that the phones are set up properly for your users:

For information on changing the language that is displayed on the User Options web page, see *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*.

Unified Communications Manager Endpoints Locale Installer

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

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

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



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



APPENDIX **C**

Technical Specifications

- [Physical and Operating Environment Specifications, on page 115](#)
- [Cable Specifications, on page 117](#)
- [Network and Access Port Pinouts, on page 117](#)
- [Phone Behavior During Times of Network Congestion, on page 119](#)

Physical and Operating Environment Specifications

The following sections describe the physical and operating environment specifications for Cisco Unified IP Phone 6901 and Cisco Unified IP Phone 6911.

Physical and Operating Environment Specifications for Cisco Unified IP Phone 6901

The following table shows the physical and operating environment specifications for the Cisco Unified IP Phone 6901.

Table 22: Physical and Operating Environment Specifications for the Cisco Unified IP Phone 6901

| Specification | Value or Range |
|-----------------------------|--|
| Operating temperature | 23° to 113°F (-5° to 45°C) |
| Operating relative humidity | 10% to 95% (noncondensing) |
| Storage temperature | -13° to 158°F (-25° to 70°C) |
| Height | 8.1 in. (20.5 cm) |
| Width | 3.7 in. (9.4 cm) |
| Depth | <ul style="list-style-type: none">• 1.8 in. (4.6 cm) - With footstand closed• 3.42 in. (8.7 cm) - With footstand open |

| Specification | Value or Range |
|-----------------------|--|
| Weight | <ul style="list-style-type: none"> • 1.37 lb (621.2 g) - Black handset phone (Europe) • 1.37 lb (662 g) - White handset phone (Europe) • 1.44 lb (655.2 g) - Black handset phone (North America) • 1.53 lb (695 g) - White handset phone (North America) |
| Power | <ul style="list-style-type: none"> • 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 for 10-Mbps cables with four pairs Category 5/5e for 100-Mbps cables with four pairs Note Cables have four pairs of wires for a total of eight conductors. |
| Distance Requirements | As supported by the Ethernet Specification, it is assumed that the maximum cable length between each Cisco Unified IP Phone and the switch is 100 meters (330 feet). |

Physical and Operating Environment Specifications for Cisco Unified IP Phone 6911

The following table shows the physical and operating environment specifications for the Cisco Unified IP Phone 6911.

Table 23: Physical and Operating Environment Specifications for the Cisco Unified IP Phone 6911

| Specification | Value or Range |
|-----------------------------|---|
| Operating temperature | 32° to 104°F (0° to 40°C) |
| Operating relative humidity | 10% to 95% (non-condensing) |
| Storage temperature | 14° to 140°F (-10° to 60°C) |
| Height | 8.1 in. (20.5 cm) |
| Width | 7.4 in. (18.8 cm) |
| Depth | 1.5 in. (3.82 cm) - Excluding the handset |

| Specification | Value or Range |
|-----------------------|---|
| Weight | Charcoal: Standard: 32.3 oz (917g); slimline: 31.1 oz (883g) Arctic white: Standard: 34.6 oz (981.3g); slimline: 33.5 oz (948.3g) |
| Power | <ul style="list-style-type: none"> • 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 for 10-Mbps cables with 4 pairs Category 5/5e for 100-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, it is assumed that the maximum cable length between each Cisco Unified IP Phone and the switch is 100 meters (330 feet). |

Cable Specifications

- RJ-9 jack (4-conductor) for handset connection.
- RJ-45 jack for the LAN 10/100BaseT connection (labeled 10/100 SW on the Cisco Unified IP Phone 6901 and 6911).
- RJ-45 jack for a second 10/100BaseT compliant connection (labeled 10/100 PC on the Cisco Unified IP Phone 6911).
- 48-volt power connector.

Network and Access Port Pinouts

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

- The network port is labeled **network** on the Cisco Unified IP Phone.
- The access port is labeled **Computer** on the Cisco Unified IP Phone (Cisco Unified IP Phone 6911 only).

Network Port Connector

The following table describes the network port connector pinouts.

Table 24: 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 25: 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 Behavior During Times of Network Congestion

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

- Administrative tasks, such as an internal port scan or security scan
- Attacks that occur on your network, such as a Denial of Service attack



APPENDIX **D**

Basic Phone Administration Steps

- [Phone Administration Overview](#), on page 121
- [Example User Information](#), on page 121
- [Cisco Unified Communications Manager User Addition](#), on page 122
- [Phone Setup](#), on page 123
- [Perform Final End User Configuration Steps](#), on page 126

Phone Administration Overview

This appendix provides minimum, basic configuration steps for you to do the following actions:

- Add a new user to Cisco Unified Communications Manager
- Configure a new phone for that user
- Associate that user to that phone
- Complete other basic end-user configuration tasks

The procedures provide one method for performing these tasks and are not the only way to perform these tasks. They are a streamlined approach to get a new user and the corresponding phone running on the system.

These procedures are designed to be used on a mature Cisco Unified Communications Manager system where calling search spaces, partitions, and other complicated configuration have already been done and are in place for existing users.

Example User Information

In the procedures that follow, examples are given when possible to illustrate some of the steps. Sample user and phone information used throughout these procedures includes:

- User's Name: John Doe
- User ID: johndoe
- Phone model: 6901
- Protocol: SCCP

- MAC address listed on phone: 00127F576611
- Five-digit internal telephone number: 26640

Cisco Unified Communications Manager User Addition

This section describes steps for adding a user to Cisco Unified Communications Manager. Follow one of the procedures in this section, depending on your operating system and the manner in which you are adding the user.

Add User from External LDAP Directory

If you added a user to an LDAP Directory (a non-Cisco Unified Communications Server directory), you can immediately synchronize that directory to the Cisco Unified Communications Manager on which you are adding this same user and the phone user by following these steps:

Procedure

- Step 1** Login to Cisco Unified Communications Manager.
- Step 2** Choose **System > LDAP > LDAP Directory**.
- Step 3** Use the **Find** button to locate your LDAP directory.
- Step 4** Click the **LDAP directory name**.
- Step 5** Click **Perform Full Sync Now**.

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

- Step 6** Proceed to [Phone Setup, on page 123](#).
-

Add User Directly to Cisco Unified Communications Manager

If you are not using an LDAP directory, you can add a user directly to Cisco Unified Communications Manager by following these steps:

Procedure

- Step 1** Choose **User Management > End User**.
- Step 2** Click **Add New**.
The End User Configuration window appears.
- Step 3** In the User Information pane of this window, enter the following:

- **User ID:** Enter the 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 user password. You may use the following special characters: =, +, <, >, #, ;, \, , ""', and blank spaces.
- **Last Name:** Enter the user last name. You may use the following special characters: =, +, <, >, #, ;, \, , ""', and blank spaces.

Example: doe

- **Telephone Number:** Enter the primary directory number for the user. Users can have multiple lines on their phones.

Example: 26640 (John Doe's internal company telephone number)

Step 4 Click **Save**.

Step 5 Proceed to the section [Phone Setup, on page 123](#).

Phone Setup

To identify the user phone model and protocol, follow these steps:

Procedure

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

Step 2 Click **Add New**.

Step 3 Select the user phone model from the Phone Type drop-down list, then click **Next**.

The Phone Configuration window appears.

Note On the Phone Configuration window, you can use the default values for most of the fields.

Step 4 For the required fields, possible values can be configured as follows:

Note The configuration is based on the example of user *johndoe*.

a) In the Device Information pane of this window:

MAC Address

Enter the MAC address of the phone, listed on a sticker on the back of phone.

The MAC address is 12 hexadecimal characters long.

Example: 00127F576611 (MAC address on John Doe's phone)

Description

This is an optional field where you can enter a useful description. This will help you if you need to search for information about this user.

Device Pool

Choose the device pool for which you want to assign this phone. The device pool defines sets of common characteristics for devices, such as region, date/time group, and MLPP information.

Note Device Pools are defined on the Device Pool Configuration window of Cisco Unified Communications Server Administration (**System > Device Pool**).

Phone Button Template

Choose the appropriate phone button template from the drop-down list. The phone button template determines the configuration of features on a phone and identifies the predetermined number to be dialed after pressing the feature button. To configure the Predetermined Number for a feature, navigate to the Phone Button Template of Cisco Unified IP Phone 6911. In the template, 1 is always assigned to the line, so the button number minus 1 is the predetermined number that is provided by your system administrator.

Note Phone button templates are defined on the Phone Button Template Configuration window of Cisco Unified Communications Manager (**Device > Device Settings > Phone Button Template**). You can use the search fields and the **Find** button to find all configured phone button templates and their current settings.

Common Phone Profile

From the drop-down list, choose a common phone profile from the list of available common phone profiles.

Note Common Phone Profiles are defined on the Common Phone Profile Configuration window of Cisco Unified Communications Manager (**Device > Device Settings > Common Phone Profile**). You can use the search fields and the **Find** button to find all configured common phone profiles and their current settings.

Calling Search Space

From the drop-down list, choose the appropriate calling search space (CSS). A calling search space comprises a collection of partitions (analogous to a collection of available phone books) that are searched to determine how a dialed number should be routed. The calling search space for the device and the calling search space for the directory number get used together. The directory number CSS takes precedence over the device CSS.

Note Calling Search Spaces are defined on the Calling Search Space Configuration window of Cisco Unified Communications Manager (**Calling routing > Class of Control > Calling Search Space**). You can use the search fields and the **Find** button to find all configured Calling Search Spaces and their current settings.

Location

Choose the appropriate location for this Cisco Unified IP Phone.

Owner User ID

From the drop-down list, choose the user ID of the assigned phone user.

- b) In the Protocol Specific Information pane of this window, choose a Device Security Profile from the drop-down list. To enable security features for a phone, you must configure a new security profile for the

device type and protocol and apply it to the phone. If the phone does not support security, choose a nonsecure profile.

To identify the settings that are contained in the profile, choose **System > Security Profile > Phone Security Profile**.

Note The security profile chosen should be based on the overall security strategy of the company.

- c) In the Extension Information pane of this window, check the Enable Extension Mobility box if this phone supports Cisco Extension Mobility.
- d) Click **Save**.

Step 5 Configure line settings:

- a) On the Phone Configuration window, click Line 1 on the left pane of the window. The Directory Number Configuration window appears.
- b) In the Directory Number field, enter a valid number that can be dialed.

Note This field should contain the same number that appears in the Telephone Number field on the User Configuration window.

Example: 26640 is the directory number of user John Doe in the example above.

- c) From the Route Partition drop-down list, choose the partition to which the directory number belongs. If you do not want to restrict access to the directory number, choose <None> for the partition.
- d) From the Calling Search Space drop-down list (Directory Number Settings pane of the Directory Number Configuration window), choose the appropriate calling search space. A calling search space comprises a collection of partitions that are searched for numbers that are called from this directory number. The value that you choose applies to all devices that are using this directory number.
- e) In the Call Pickup and Call Forward Settings pane of the Directory Number Configuration window, choose the items (for example, Forward All, Forward Busy Internal) and corresponding destinations to which calls should be sent.

Example: If you want incoming internal and external calls that receive a busy signal to be forwarded to the voice mail for this line, check the Voice Mail box next to the Forward Busy Internal and Forward Busy External items in the left column of the Call Pickup and Call Forward Settings pane.

- f) In the Line 1 on Device... pane of the Directory Number Configuration window, configure the following parameters:

Display (Internal Caller ID field)

You can enter the first name and last name of the user of this device so that this name will be displayed for all internal calls. You can also leave this field blank to have the system display the phone extension.

External Phone Number Mask

Indicate phone number (or mask) that is used to send Caller ID information when a call is placed from this line.

You can enter a maximum of 24 number and “X” characters. The X characters represent the directory number and must appear at the end of the pattern.

Example: Using the John Doe extension in the example above, if you specify a mask of 408902XXXX, an external call from extension 6640 displays a caller ID number of 4089026640.

This setting applies only to the current device unless you check the Update Shared Device Settings check box at right and click the **Propagate Selected** button. The check box only displays if other devices share this directory number.

- g) Click **Save**.
- h) Click **Associate End Users** at the bottom of the window to associate a user to the line being configured.
- i) Use the **Find** button and the Search fields to locate the user, check the box next to the user name, and then click **Add Selected**.

The user's name and user ID should appear in the Users Associated With Line pane of the Directory Number Configuration window.

- j) Click **Save**. The user is now associated with Line 1 on the phone.
- k) If the phone has a second line, configure Line 2.
- l) Associate the user with the device:
 1. Choose **User Management > End User**.
 2. Use the search boxes and the **Find** button to locate the user you have added (for example, doe for the last name).
 3. Click on the user ID (for example, *johndoe*). The End User Configuration window appears.
 4. Click **Device Associations**.
 5. Use the Search fields and the **Find** button to locate the device with which you want to associate to the user.
 6. Select the device, then click **Save Selected/Changes**. The user is now associated with the device.
 7. Click the **Go** button next to the Back to User Related link in the upper-right corner of the screen.

Step 6 Proceed to [Perform Final End User Configuration Steps, on page 126](#).

Perform Final End User Configuration Steps

If you are not already on the End User Configuration page, choose **User Management > End User** to perform some final configuration tasks. Use the Search fields and the **Find** button 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. In the **End User configuration** window, do the following:

Procedure

- Step 1** In the Directory Number Associations pane of the screen, set the primary extension from the drop-down list.
- Step 2** In the Mobility Information pane, check the Enable Mobility box.
- Step 3** In the Permissions Information pane, 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 has been defined as a Standard CCM End User Group.
To view all configured user groups, choose **User Management > User Group**.

Step 4 Click **Save**.



APPENDIX E

Cisco Unified IP Phone Wall Mount

- [Wall Mount Components for Cisco Unified IP Phone 6901, on page 129](#)
- [Wall Mount Components for Cisco Unified IP Phone 6911, on page 132](#)
- [Adjust Handset Rest, on page 137](#)

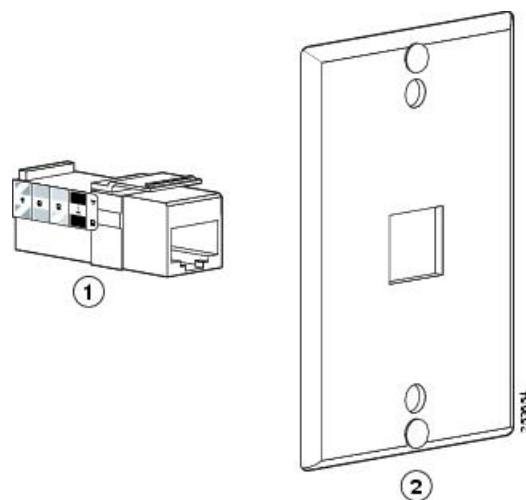
Wall Mount Components for Cisco Unified IP Phone 6901

The following section describes how to install the Wall Mount plate for use with the Cisco Unified IP Phone 6901.

Cisco recommends that you use the Leviton Wall Mount plate (Leviton type number: 4108W-OSP) for mounting the Cisco Unified IP Phone 6901. The wall mount plate must be ordered separately from the phone.

The following figure and table show a list of items required to mount the Cisco Unified IP Phone 6901 on the wall using the Leviton Wall Mount plate.

Figure 1: Leviton Wall Mount plate



| | | | |
|---|----------------|---|--------------------------|
| 1 | RJ45 Connector | 2 | Leviton Wall Mount Plate |
|---|----------------|---|--------------------------|

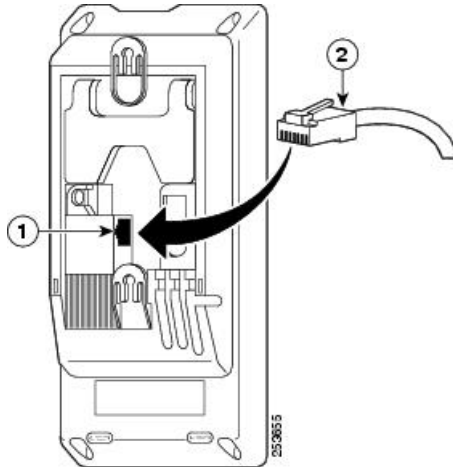
Install Phone on Wall Mount Plate

To install the phone on the wall mount plate, perform these steps:

Procedure

- Step 1** Plug the telephone line cord (RJ45 connector) into the phone jack at the base of the phone.

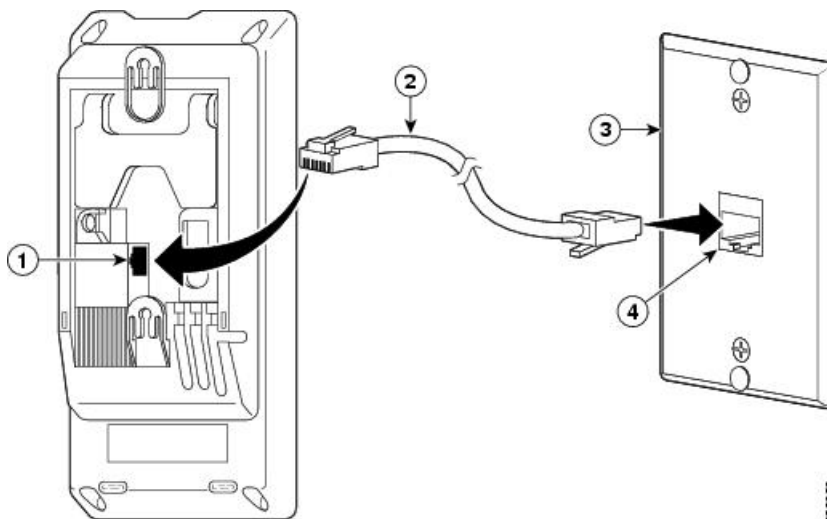
Figure 2: RJ45 Connector in the Phone Jack



| | |
|---|---------------------------|
| 1 | Network Port on the Phone |
| 2 | RJ45 Connector |

- Step 2** Plug the RJ45 connector into the wall mount phone jack.

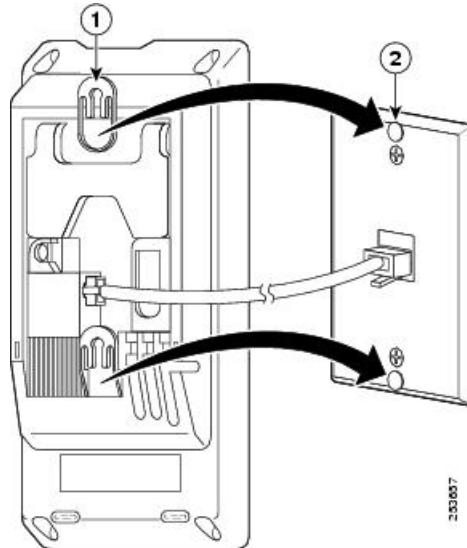
Figure 3: RJ45 Connector in the Wall Mount Jack



| | | | |
|---|---------------------------|---|--------------------------------------|
| 1 | Network Port on the Phone | 3 | Wall Mount Plate |
| 2 | RJ45 Connector | 4 | Network Port on the Wall Mount Plate |

Step 3 Slip the mounting holes on the base of the wall mount plate and over the wall mount pins.

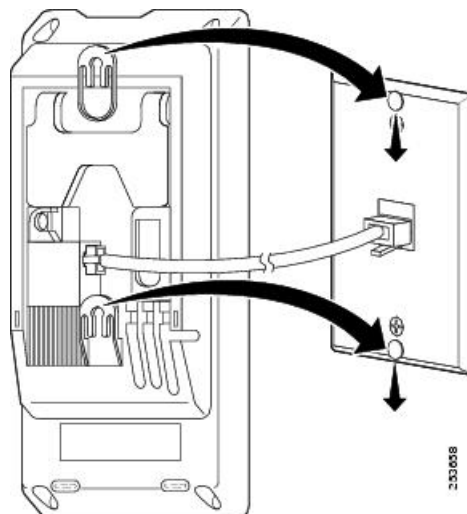
Figure 4: Mounting Holes



| | |
|---|--|
| 1 | Mounting Hole on the Phone |
| 2 | Wall Mount Pin on the Wall Mount Plate |

Step 4 Firmly slide the IP Phone down into place.

Figure 5: Sliding the IP Phone



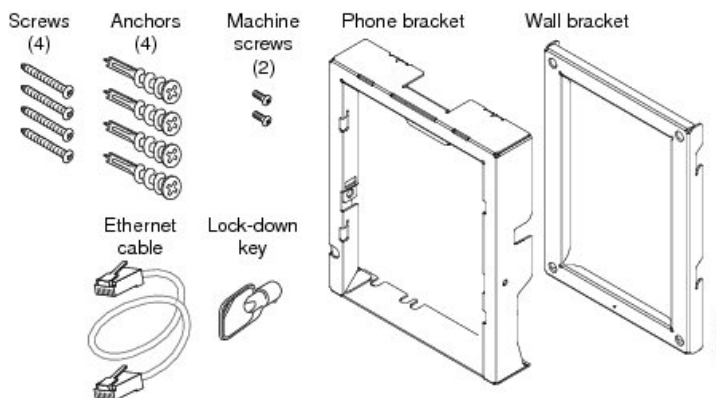
Step 5 Change the handset as described in [Adjust Handset Rest, on page 137](#).

Wall Mount Components for Cisco Unified IP Phone 6911

The following section describes how to install the wall mount for use with the Cisco Unified IP Phone 6911.

The following figure shows a list of items included in the Wall Mount Kit for the Cisco Unified IP Phone 6911.

Figure 6: Wall Mount Kit for Cisco Unified IP Phone 6911



The package includes the following items:

- One phone bracket
- One wall bracket
- Four #10-12x1-inch Phillips-head screws with 4 anchors
- Two #4-40x1/4-inch machine screws
- One 6-inch Ethernet cable
- One key if the bracket includes the optional lock

Before You Begin

You need these tools to install the bracket:

- #1 and #2 Phillips-head screwdrivers
- Level

You must also install an Ethernet jack for the telephone in the desired location if an Ethernet jack does not currently exist. This jack must be wired appropriately for an Ethernet connection. You cannot use a regular telephone jack. For more information on phone installation requirements and warnings, see [Cisco Unified IP Phone Installation, on page 41](#).

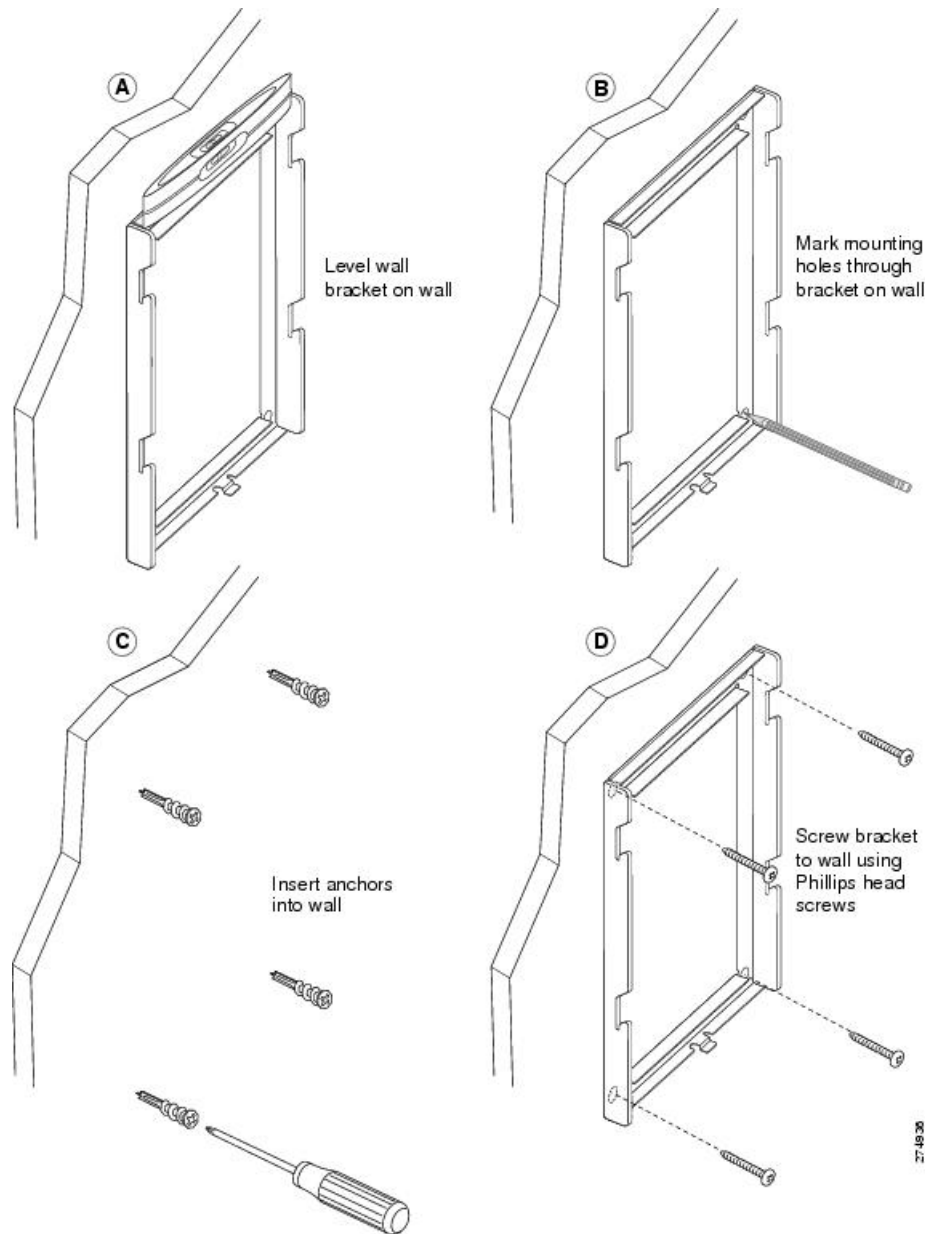
Install Bracket

To install the phone on the wall, perform these steps:

Procedure

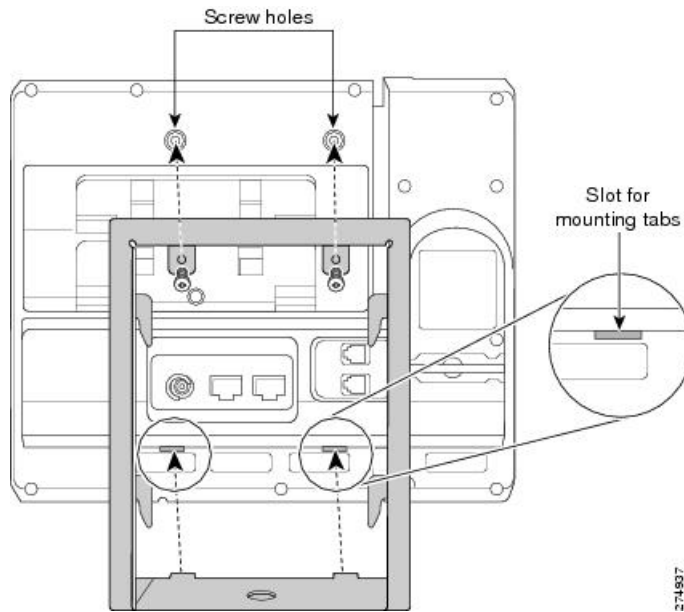
- Step 1** Mount the wall bracket in the desired location. You can install the wall bracket over an Ethernet jack, or you can run the Ethernet network cable to a jack nearby.
- a) Use the level to ensure the wall bracket is level, then use a pencil to mark the screw holes.
 - b) Using a #2 Phillips-head screwdriver, carefully center the anchor over the pencil mark and press the anchor into the wall to make markings on the wall.
 - c) Screw the anchor clockwise into the wall until it is seated flush.
 - d) Use the included screws and a #2 Phillips-head screwdriver to attach the wall bracket to the wall.

Figure 7: Mounting the Wall Bracket

**Step 2** Attach the phone bracket to the IP Phone.

- Detach the handset cord (and headset cord, if there is a headset), power cord, and any other attached cords from the base of the phone.
- Remove the label covers that are concealing the screw holes.
- Attach the phone bracket by inserting the tabs into the mounting tabs on the phone. The phone ports should be accessible through the holes in the bracket.
- Secure the phone bracket to the IP Phone with the Phillips-head screws.
- Thread the handset cord (and headset cord, if using one). Reattach the cords and seat them in the clips incorporated into the phone body.

Figure 8: Attaching the Phone Bracket

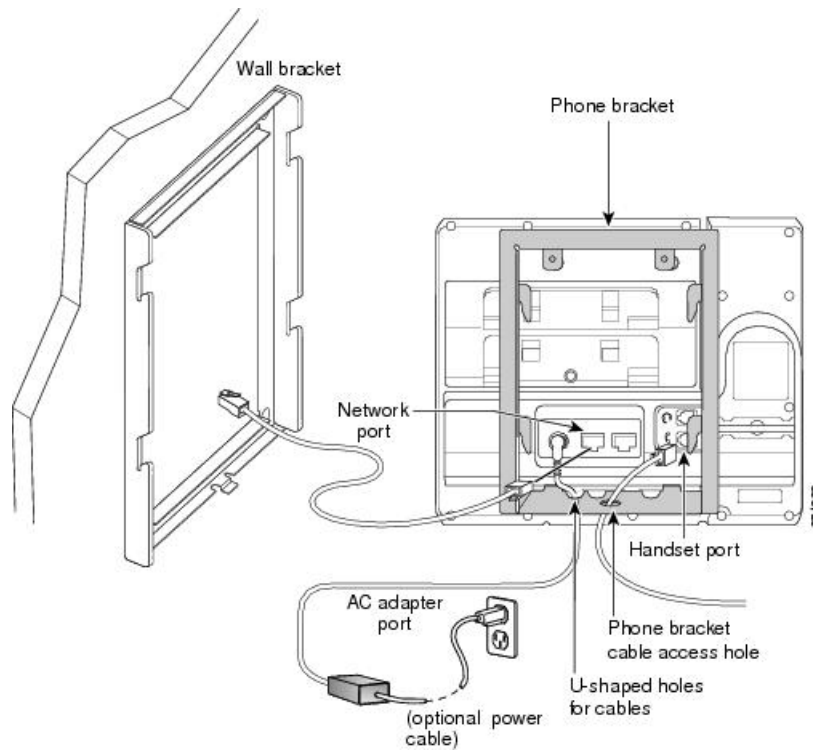


Step 3 Attach the Ethernet cable to the 10/100 SW network port and wall jack.

If you are connecting a network device (such as a computer) to the phone, attach the cable to the 10/100 PC access port.

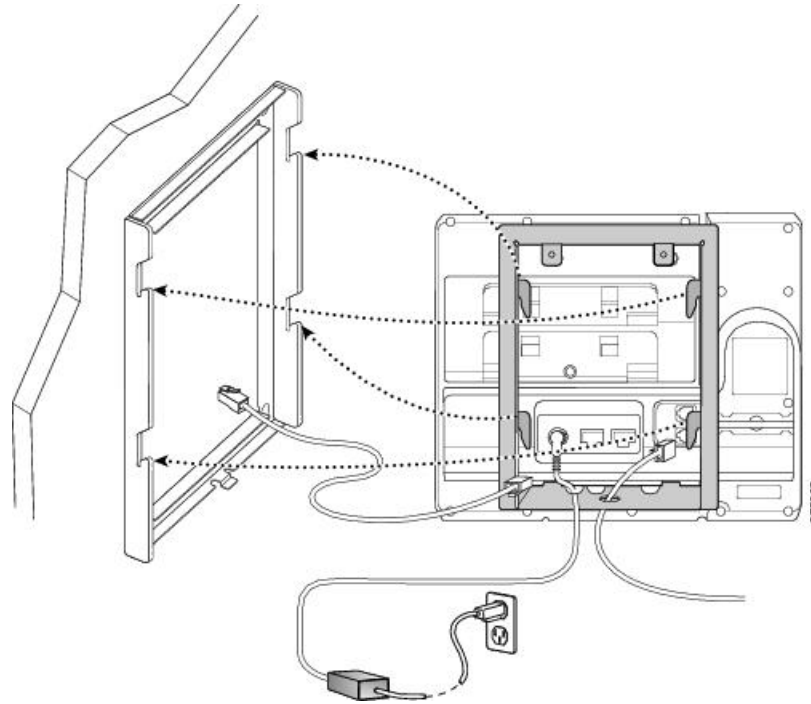
If you are using an external power supply, plug the power cord into the phone and dress the cord by clipping it into the clips incorporated into the phone body next to the 10/100 PC port.

Figure 9: Attaching the Cables



- Step 4** Attach the phone to the wall bracket by inserting the tabs on the top of the phone bracket into the slots on the wall bracket. Ensure that the power cord and any other cable that does not terminate in the wall behind the bracket are positioned in one of the cable-access openings in the bottom of the bracket. The phone and wall bracket openings together form circular openings with room for one cable per opening.
- Step 5** Use the locking key to lock the phone to the wall bracket.

Figure 10: Attaching the Phone to the Wall Bracket

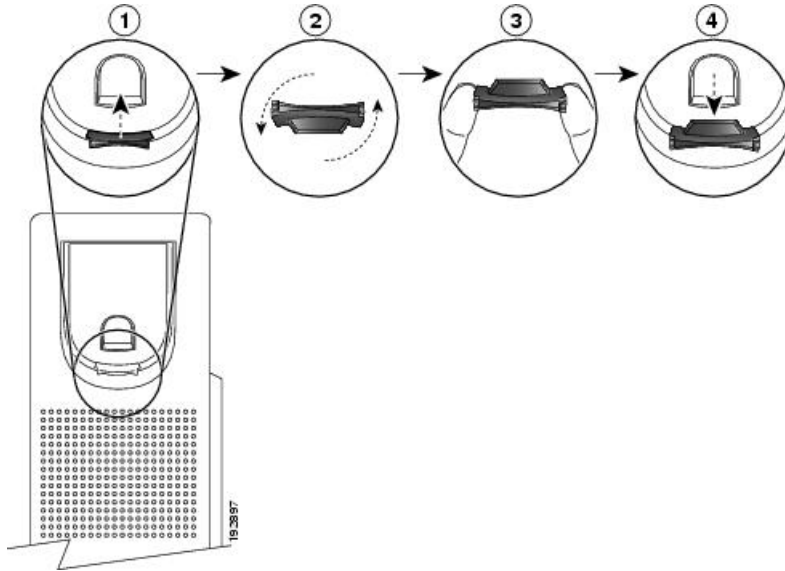


Step 6 Change the handset as described in [Adjust Handset Rest, on page 137](#).

Adjust Handset Rest

With a wall-mounted phone, you might need to adjust the handset rest to ensure that the receiver does not slip out of the cradle. The hook should have a lip on which the handset catches when the phone is vertical. Follow the diagram and steps below to change the hookswitch hook.

Figure 11: Adjust the Handset Hook



| | |
|---|--|
| 1 | Remove the handset from the cradle and pull the plastic tab from the handset rest. |
| 2 | Rotate the tab 180 degrees. |
| 3 | Hold the tab between two fingers, with the corner notches facing you. |
| 4 | Line up the tab with the slot in the cradle, and press the tab evenly into the slot. An extension protrudes from the top of the rotated tab. Return the handset to the handset rest. |



APPENDIX **F**

Cisco Unified IP Phone Non-Lockable Wall Mount

- [ADA Non-Lockable Wall Mount Kit for 6900 Series, on page 139](#)

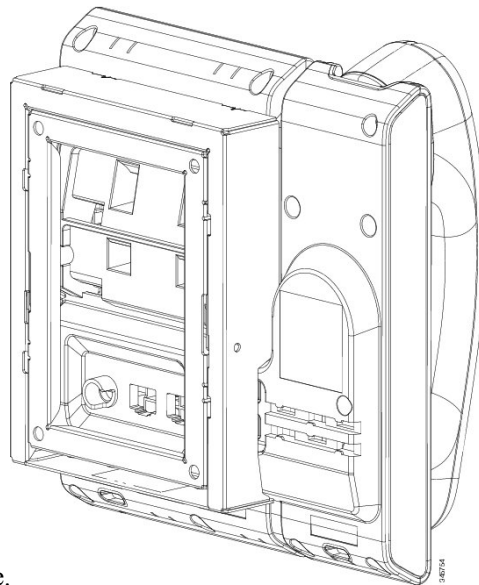
ADA Non-Lockable Wall Mount Kit for 6900 Series

This section describes how to install the ADA Non-Lockable Wall Mount Kit for 6900 Series on a Cisco Unified IP Phone 6911, 6921, 6941, 6945, and 6961.

The nonlocking wall mount kit meets ADA 4.4.1 requirements.

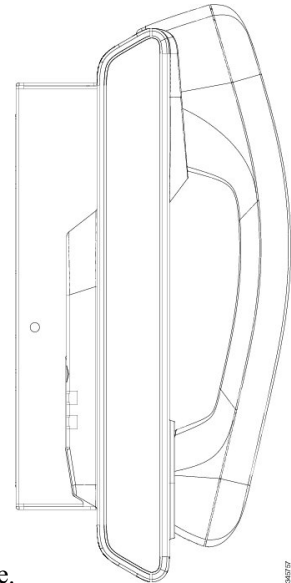
Figure 12: Back View of ADA Non-Lockable Wall Mount Kit Installed on Phone

The following figure shows the wall mount kit installed on the



phone.

Figure 13: Side View of ADA Non-Lockable Wall Mount Kit Installed on Phone

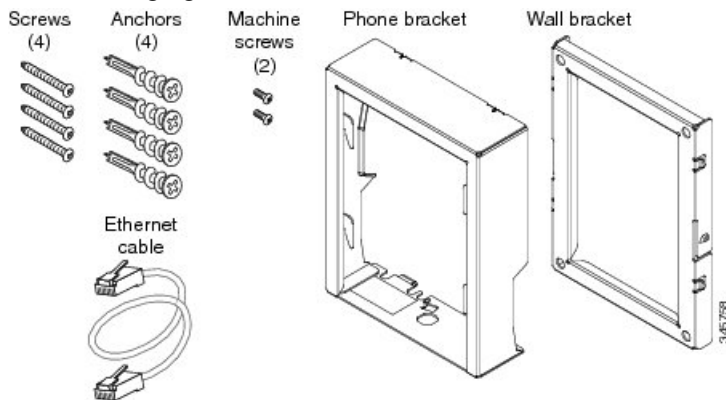


The following figure shows the phone with the wall mount kit from the side.

Components

Figure 14: Components

The following figure shows the contents of the Wall Mount kit.



The package includes these items:

- One phone bracket
- One wall bracket
- Four #8-18 x 1.25-inch Phillips-head screws with four anchors
- Two M2.5 x 6 mm machine screws
- One 6-inch Ethernet cable

Before You Begin

You need these tools to install the bracket:

- #1 and #2 Phillips-head screwdrivers
- Level
- Pencil

You must also install an Ethernet jack for the telephone in the desired location if an Ethernet jack does not currently exist. This jack must be wired appropriately for an Ethernet connection. You cannot use a regular telephone jack.

Related Topics

[Cisco Unified IP Phone Installation](#), on page 41

Install Non-Lockable Wall Mount Kit for Phone

The wall mount kit can be mounted on most surfaces, including concrete, brick, and similar hard surfaces. To mount the kit on concrete, brick, or similar hard surfaces, you must provide the appropriate screws and anchors for your wall surface.

Procedure

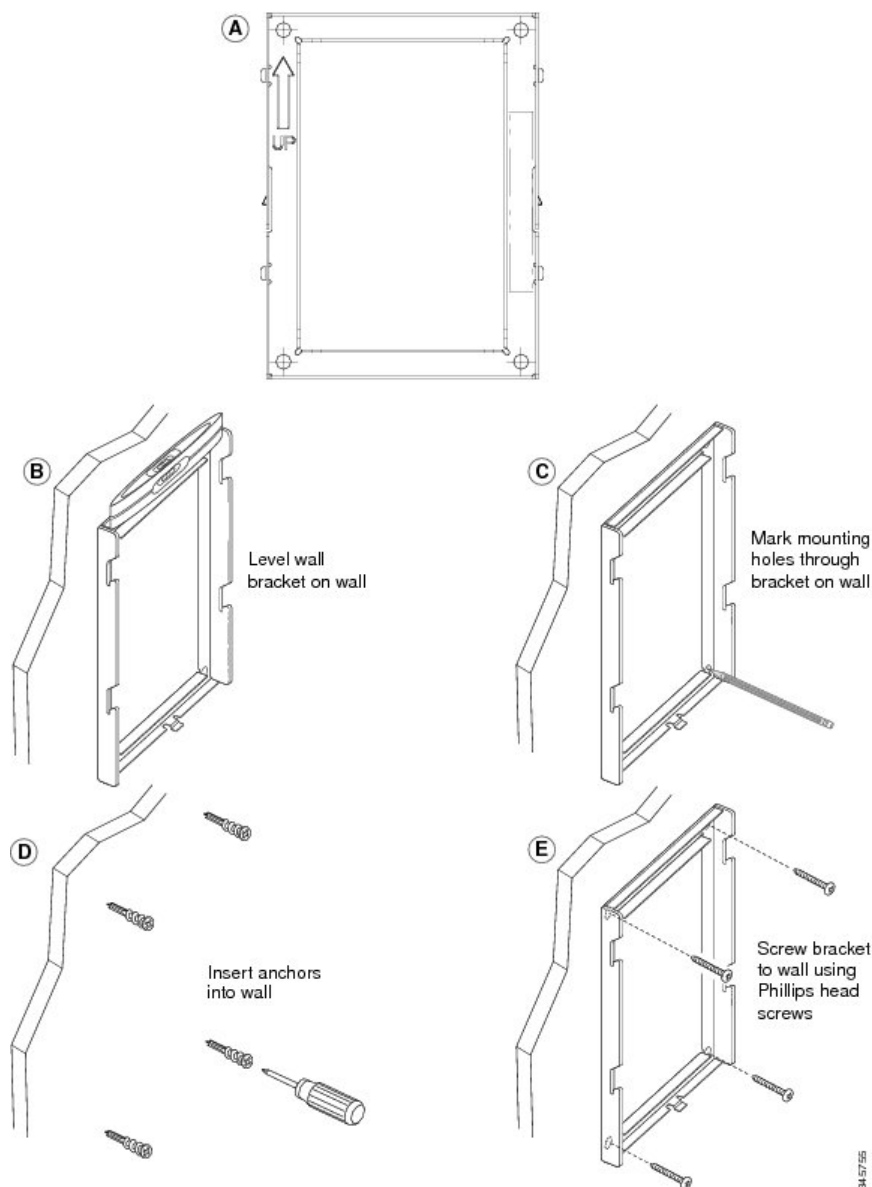
Step 1 Mount the wall bracket in the desired location. You can install the bracket over an Ethernet jack, or you can run the Ethernet network cable to a jack nearby.

Note If the jack is to be placed behind the phone, the Ethernet jack must be flush to the wall or recessed.

- Hold the bracket on the wall, placing it so that the arrow on the back of the bracket is pointing up.
- Use the level to ensure the bracket is level and use a pencil to mark the screw holes.
- Use a #2 Phillips-head screwdriver to carefully center the anchor over the pencil mark and press the anchor into the wall.
- Screw the anchor clockwise into the wall until it is seated flush.
- Use the included screws and a #2 Phillips-head screwdriver to attach the bracket to the wall.

Figure 15: Mount Wall Bracket

The following figure shows the steps to mount the wall bracket.

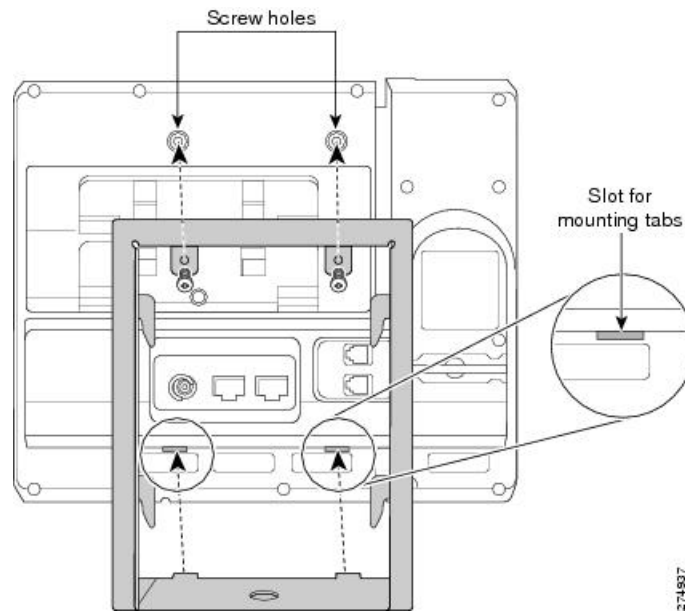


Step 2 Attach the phone bracket to the IP phone.

- Detach the handset cord (and headset cord, if there is a headset), power cord, and any other attached cords from the base of the phone.
- Remove the label covers that are concealing the screw holes.
- Attach the phone bracket by inserting the tabs into the mounting tabs on the phone. The phone ports should be accessible through the holes in the bracket.
- Secure the phone bracket to the IP Phone with the machine screws using the #1 Phillips-head screwdriver.
- Thread the handset cord (and headset cord, if using one). Reattach the cords and seat them in the clips incorporated into the phone body.

Figure 16: Attach Phone Bracket

The following figure shows how to attach the phone bracket.

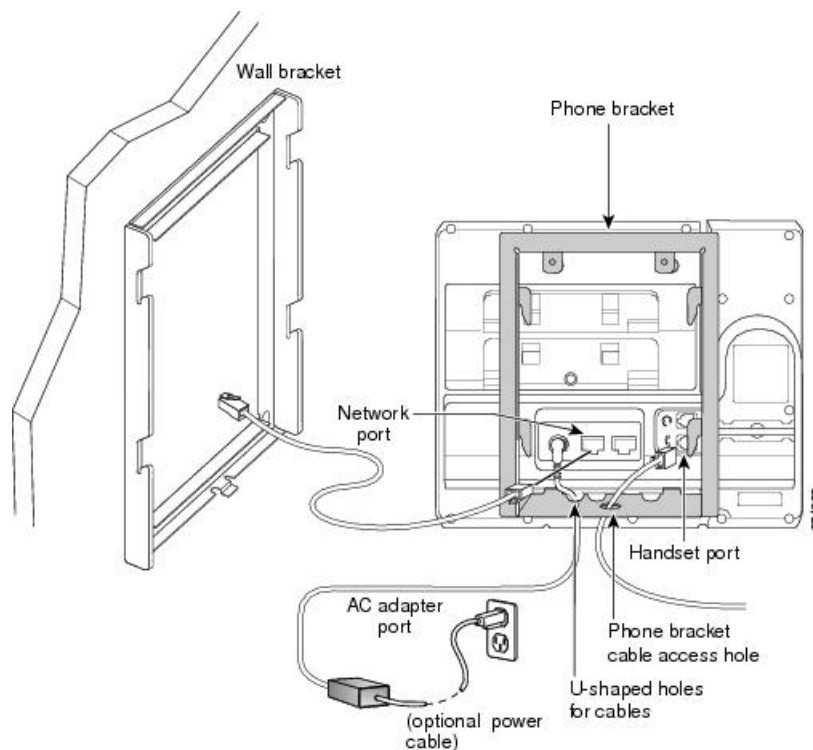


Step 3 Attach the cables.

- a) Attach the Ethernet cable to the 10/100 SW network port and wall jack.
- b) (Optional) If you are connecting a network device (such as a computer) to the phone, attach the cable to the 10/100 PC access port.
- c) (Optional) If you are using an external power supply, plug the power cord into the phone and dress the cord by clipping it into the clips incorporated into the phone body next to the 10/100 PC port.
- d) (Optional) If the cables terminate inside the wall bracket, connect the cables to the jacks.

Figure 17: Attach Cables

The following figure shows the cable attachment.



Step 4 Attach the phone to the wall bracket by inserting the tabs on the top of the phone bracket into the slots on the wall bracket.

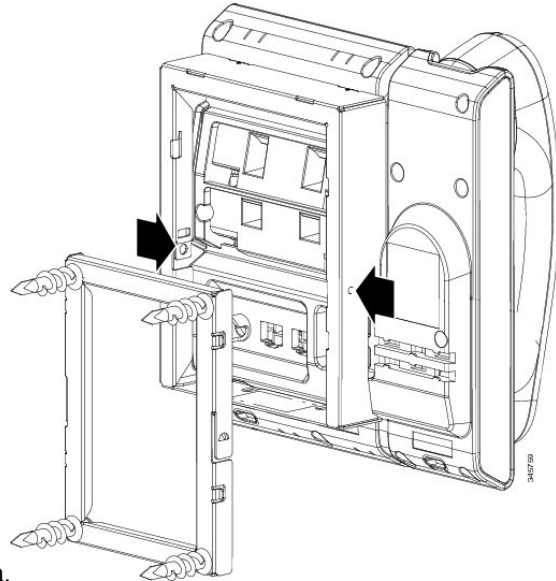
For cables that terminate outside of the bracket, use the cable-access openings in the bottom of the bracket to position the power cord and any other cable that does not terminate in the wall behind the bracket. The phone and wall bracket openings together form circular openings with room for one cable per opening.

Step 5 Proceed to [Adjust Handset Rest](#), on page 137.

Remove Phone from Non-Lockable Wall Mount

The phone mounting plate contains two tabs to lock the plate into the wall bracket. The following figure shows the location and shape of the tabs.

Figure 18: Tab Location



The following figure shows the tab location.

To remove the phone and mounting plate from the wall bracket, you must disengage these tabs.

Before you begin

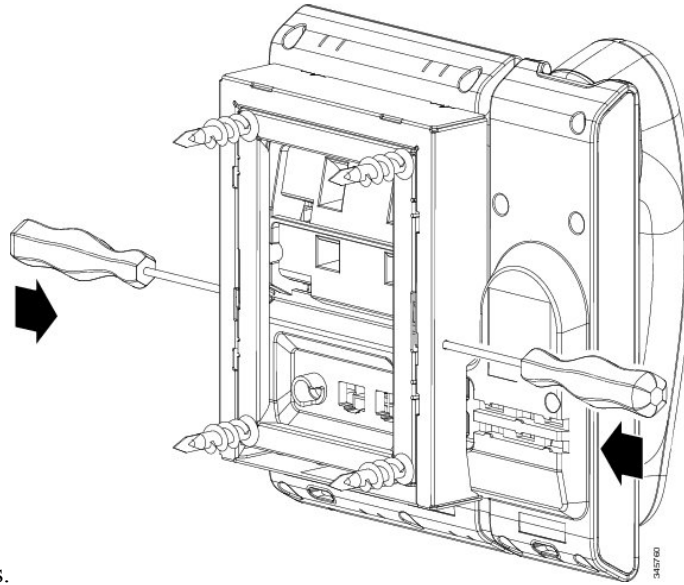
You require two screwdrivers or metal rods.

Procedure

-
- Step 1** Push the screw drivers into the left and right holes in the phone mounting plate approximately 1 in. (2.5 cm).
- Step 2** Press firmly inwards (towards the phone) to disengage the tabs, lift up on the phone to release the phone from the wall bracket, and then pull the phone towards you.

Figure 19: Disengage Tabs

The following figure shows how to disengage the



tabs.



APPENDIX G

Feature Support by Protocol

- [Feature Support, on page 147](#)

Feature Support

This appendix provides information about feature support for the Cisco Unified IP Phones 6901 and 6911 using the SCCP or SIP protocol with Cisco Unified Communications Manager.

The following table provides a high-level overview of calling features and their support by protocol. This table focuses primarily on end-user calling features and is not intended to represent a comprehensive listing of all available phone features. For details about user interface differences and feature use, refer to the *Cisco Unified IP Phones 6901 and 6911 User Guide for Cisco Unified IP Phones 6901 and 6911 (SCCP and SIP)*.

The guide is available at this URL:

http://www.cisco.com/en/US/products/ps10326/tsd_products_support_series_home.html

The specific sections that describe the features in the user guide are referenced in the following table.

Table 26: Cisco Unified IP Phones 6901 and 6911 feature support by protocol

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|---|----------------|---------------|---|
| Abbreviated Dialing | Not supported | Not supported | |
| Agent Greeting | Supported | Supported | |
| Assisted Directed Call Park | Not supported | Not supported | |
| Audible Message Waiting Indicator (AMWI) | Supported | Supported | Calling features - Messages |
| Auto Answer (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Auto Answer |

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|--|-----------------------|----------------------|--|
| Automatic Port Synchronization (Cisco Unified IP Phone 6911 only) | Supported | Supported | |
| cBarge | Supported | Supported | Calling features - Shared lines |
| Block external to external transfer | Supported | Supported | |
| Busy Lamp Field (BLF) | Not supported | Not supported | |
| Busy Lamp Field (BLF) Pickup | Not supported | Not supported | |
| Call Back | Supported | Supported | |
| Call Display Restrictions | Not supported | Not supported | |
| Call Forward All | Supported | Supported | Calling features - Call Forward |
| Call Forward All Breakout | Supported | Supported | Calling features - Call Forward |
| Call Forward All Loop Prevention | Supported | Supported | Calling features - Call Forward |
| Call Forward Busy | Supported | Supported | Calling features - Call Forward |
| Call Forward Configurable Display | Supported | Supported | Calling features - Call Forward |
| Call Forward Destination Override | Supported | Supported | Calling features - Call Forward |
| Call Forward No Answer | Supported | Supported | Calling features - Call Forward |
| Call History for Shared Line | Not supported | Not supported | |
| Call Park | Not supported | Not supported | |

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|--|---------------------|---------------------|---|
| Call Pickup Group Call Pickup (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Call Pickup |
| Call Recording | Not supported | Not supported | |
| Call Waiting | Supported | Supported | Calling features - Call Waiting |
| Call Waiting Ring | Not supported | Not supported | |
| Caller ID | Not supported | Not supported | |
| Caller ID Blocking | Not supported | Not supported | |
| Calling Party Normalization | Not supported | Not supported | |
| Cisco Extension Mobility | Not supported | Not supported | |
| Cisco Extension Mobility Cross Cluster | Not supported | Not supported | |
| Client Matter Code (CMC) | Supported | Not supported | Calling features - Codes |
| Computer Telephony Integration (CTI) Applications | Not supported | Not supported | |
| Configurable Call Forward Display | Not supported | Not supported | |
| Device Invoked Recording | Not supported | Not supported | |
| Direct Transfer | Not supported | Not supported | |
| Directed Call Park | Not supported | Not supported | |
| Directed Call Pickup | Not supported | Not supported | |
| Disable Single Button Barge | Not supported | Not supported | |
| Do Not Disturb (DND) | Not supported | Not supported | |
| Distinctive Ring | Partially supported | Partially supported | |

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|---|-----------------------|----------------------|--|
| Electronic Hookswitch | Not supported | Not supported | |
| Enhanced Secure Extension Mobility Cross Cluster | Not supported | Not supported | |
| Enhanced Version Negotiation with Cisco Unified Manager Express | Not supported | Supported | |
| EnergyWise | Supported | Supported | Features of your Cisco Unified IP Phone - Power-Saving mode |
| E-SRST Service Improvements | Not supported | Not supported | |
| Fast Dial Service | Not supported | Not supported | |
| Forced Authorization Code (FAC) | Supported | Not supported | Calling features - Codes |
| Group Call Pickup (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Call Pickup |
| Headset Sidetone Control | Not supported | Not supported | |
| Hold/Resume | Supported | Supported | Calling features - Hold |
| Hold Reversion | Not supported | Not supported | |
| HTTPS | Not supported | Not supported | |
| Hunt Group | Not supported | Not supported | |
| Immediate Divert | Not supported | Not supported | |
| Incoming Call Toast Timer | Not supported | Not supported | |
| Intercom | Not supported | Not supported | |
| IPv6 Support for SIP | Not supported | Not supported | |
| Jitter Buffer | Supported | Supported | |
| Join | Not supported | Not supported | |
| Join Across Lines | Not supported | Not supported | |

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|--|-----------------------|----------------------|--|
| Line Status for Call Lists | Not supported | Not supported | |
| Log Out of Hunt Group | Not supported | Not supported | |
| Malicious Call ID | Not supported | Not supported | |
| Meet Me Conference (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Conference |
| Message Waiting Indicator | Supported | Supported | Messages |
| Minimum Ring Volume | Not supported | Not supported | |
| Mobile Connect | Not supported | Not supported | |
| Mobile Voice Access | Not supported | Not supported | |
| Monitoring and Recording (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Monitoring and Recording |
| Multilevel Precedence and Preemption (MLPP) | Supported | Not supported | Calling features - Multilevel Precedence and Preemption |
| Multiple Calls Per Line Appearance | Not supported | Not supported | |
| Music on Hold | Supported | Supported | |
| Mute (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Mute |
| Native Queuing | Not supported | Not supported | |
| No Alert Name | Not supported | Not supported | |
| On-hook Dialing | Not supported | Not supported | |
| Other Group Pickup | Not supported | Not supported | |
| Peer Firmware Sharing | Not supported | Not supported | |
| Phone Display Message for Extension Mobility Users | Not supported | Not supported | |

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|--|-----------------------|----------------------|--|
| PLK Support for Queue Statistics | Not supported | Not supported | |
| Plus Dialing | Supported | Supported | Calling features - Plus Dialing |
| Privacy | Not supported | Not supported | |
| Private Line Automated Ringdown (PLAR) | Supported | Supported | |
| Programmable Feature Buttons | Not supported | Not supported | |
| Programmable Feature Buttons as Softkeys | Not supported | Not supported | |
| PSTN Mode | Not supported | Not supported | |
| Rate Limit for Multicast and Broadcast Traffic | Not supported | Not supported | |
| Quality Reporting Tool (QRT) | Not supported | Not supported | |
| Redial | Supported | Supported | Calling features - Redial |
| Ring Setting | Not supported | Not supported | |
| Ringer Volume Control | Supported | Supported | |
| RTCP Behavior on Hold | Not supported | Supported | |
| Save Volume Change | Not supported | Not supported | |
| Secure Conference | Not supported | Not supported | |
| Secure EMCC | Not supported | Not supported | |
| Security by Default | Not supported | Not supported | |
| Serviceability for SIP Endpoints | Not supported | Not supported | |
| Services | Not supported | Not supported | |
| Services URL Button | Not supported | Not supported | |
| Shared Line | Supported | Supported | Calling features - Shared lines |

| Features | Protocol: SCCP | Protocol: SIP | For more information, see the following section in the user guide |
|---|-----------------------|----------------------|--|
| Show Call Duration in Call History | Not supported | Not supported | |
| Show Calling ID and Calling Number | Not supported | Not supported | |
| Speed Dialing (Cisco Unified IP Phone 6911 only) | Supported | Supported | Calling features - Speed Dial |
| SRST Notification | Not supported | Not supported | |
| SSH Access | Supported | Supported | |
| Time-of-Day Routing | Supported | Supported | |
| Transfer | Supported | Supported | Calling features - Transfer |
| Transfer - Direct Transfer | Not supported | Not supported | |
| TVS | Not supported | Not supported | |
| Time Zone Update | Partially supported | Partially supported | |
| UCR 2008 | Supported | Not supported | |
| Voice Mail | Supported | Supported | Messages |
| Web Access Disabled by Default | Supported | Supported | |

