



Cisco Unified IP Phones

This chapter provides information about Cisco Unified IP Phones which, as full-featured telephones, can plug directly into your IP network. H.323 clients, CTI ports, and Cisco IP Communicator represent software-based devices that you configure similarly to the Cisco Unified IP Phones. Cisco Unified Communications Manager Administration allows you to configure phone features such as call forwarding and call waiting for your phone devices. You can also create phone button templates to assign a common button configuration to a large number of phones.

After you have added the phones, you can associate users with them. By associating a user with a phone, you give that user control over that device.

- [Phone Configuration, page 2](#)
- [Supported Cisco Unified IP Phones, page 4](#)
- [Third-Party SIP Endpoints, page 25](#)
- [H.323 Clients and CTI Ports, page 26](#)
- [CTI Remote Device Setup, page 26](#)
- [Client Services Framework Setup, page 31](#)
- [Cisco IP Communicator, page 47](#)
- [Cisco Unified Personal Communicator, page 47](#)
- [Cisco TelePresence, page 48](#)
- [Cisco Unified Mobile Communicator, page 48](#)
- [Codec Usage, page 48](#)
- [Phone Button Templates, page 50](#)
- [Programmable Line Keys, page 59](#)
- [Softkey Templates, page 61](#)
- [Softkey Template Operation, page 64](#)
- [Common Phone Profiles, page 65](#)
- [Methods for Adding Phones, page 65](#)
- [Phone Migration, page 66](#)

- [Phone Features, page 67](#)
- [Phone Association, page 85](#)
- [Phone Administration Tips, page 86](#)
- [Phone Failover And Fallback, page 89](#)

Phone Configuration

Cisco Unified IP Phones, as full-featured telephones, can plug directly into your IP network. H.323 clients, CTI ports, and Cisco IP Communicator represent software-based devices that you configure similarly to the Cisco Unified IP Phones. Cisco Unified Communications Manager Administration allows you to configure phone features such as call forwarding and call waiting for your phone devices. You can also create phone button templates to assign a common button configuration to a large number of phones.

After you have added the phones, you can associate users with them. By associating a user with a phone, you give that user control over that device.

The following sections provides steps to manually configure phone that runs SCCP, and to manually configure a phone that runs SIP in Cisco Unified Communications Manager Administration. If you are using autoregistration, Cisco Unified Communications Manager adds the phone and automatically assigns the directory number.

Configure Phone For SCCP

Procedure

Step 1 Gather the following information about the phone:

- Model
- MAC address
- Physical location of the phone
- Cisco Unified Communications Manager user to associate with the phone
- Partition, calling search space, and location information, if used
- Number of lines and associated DNs to assign to the phone

[Phone Search, on page 86](#)

- Step 2** Add and configure the phone.
- Step 3** If security is required, configure the phone security profile. The phone security profile gets added to the phone by choosing a phone security profile in the Phone Configuration window.
- Step 4** If the phone will be used outside of the trusted network, configure VPN client. The VPN connection is used for situations in which a phone is located outside a trusted network or when network traffic between the phone and Cisco Unified Communications Manager must cross untrusted networks.
- Step 5** Add and configure lines (DNs) on the phone. You can also configure phone features such as call park, call forward, and call pickup.
- Step 6** Configure speed-dial buttons. You can configure speed-dial buttons for phones if you want to provide speed-dial buttons for users or if you are configuring phones that do not have a specific user who is assigned to them. Users can change the speed-dial settings on their phones by using Cisco Unified Communications Self Care Portal.
- Step 7** Configure Cisco Unified IP Phone services. You can configure services for Cisco Unified IP Phones and Cisco IP Communicator if you want to provide services for users or if you are configuring phones that do not have a specific user who is assigned to them. Users can change the services on their phones by using Cisco Unified Communications Self Care Portal.
- Step 8** Customize phone button templates and softkey templates, if required. Configure templates for each phone.
- Step 9** Configure the Busy Lamp Field feature, if required. You must use customized phone button templates to configure BLF/SpeedDial buttons.
- Step 10** Assign services to phone buttons, if required.
- Step 11** Provide power, install, verify network connectivity, and configure network settings for the Cisco Unified IP Phone.
- Step 12** Associate user with the phone (if required).
- Step 13** Make calls with the Cisco Unified IP Phone.
-

Configure Phone For SIP

The configuration steps for Cisco Unified IP Phones that support SIP are as follows.

Procedure

- Step 1** Gather the following information about the phone:
- Model
 - MAC address
 - Physical location of the phone
 - Cisco Unified Communications Manager user to associate with the phone
 - Partition, calling search space, and location information, if used
 - Number of lines and associated DN to assign to the phone

[Phone Search](#), on page 86

- Step 2** If configuring a phone that runs SIP in a secure mode, configure the SIP Phone Port in the Cisco Unified CM Configuration window.
- Step 3** If security is required, configure the phone security profile. The phone security profile gets added to the phone that runs SIP by choosing a phone security profile in the Phone Configuration window.
- Step 4** If the phone will be used outside of the trusted network, configure VPN client. The VPN connection is used for situations in which a phone is located outside a trusted network or when network traffic between the phone and Cisco Unified Communications Manager must cross untrusted networks.
- Step 5** Configure the SIP Profile. The SIP Profile gets added to the phone that runs SIP by choosing the profile in the Phone Configuration window.
- Step 6** If you are using NTP for the timing synchronization, configure the NTP server by using the Phone NTP Reference Configuration window. Add the NTP server to Date/Time Group Configuration and then assign the date/time group to the device pool. Add the device pool to the phone that runs SIP by choosing the device pool in the Phone Configuration window.
- Step 7** If you want the digits to be collected before sending them to Cisco Unified Communications Manager, configure a dial plan for the phone that runs SIP. Add the SIP Dial Rule to the phone that runs SIP by using the Phone Configuration window.
- Step 8** Add and configure the phone that runs SIP.
- Step 9** Add and configure lines (DNs) on the phone. You can also configure phone features such as call park, call forward, and call pickup.
- Step 10** Configure speed-dial buttons. You can configure speed-dial buttons for phones if you want to provide speed-dial buttons for users or if you are configuring phones that do not have a specific user who is assigned to them. Users can change the speed-dial settings on their phones by using Cisco Unified Communications Self Care Portal.
- Step 11** Configure Cisco Unified IP Phone services. You can configure services for Cisco Unified IP Phones and Cisco IP Communicator if you want to provide services for users or if you are configuring phones that do not have a specific user who is assigned to them. Users can change the services on their phones by using the Cisco Unified Communications Self Care Portal window.
- Step 12** Customize phone button templates and softkey templates, if required. Configure templates for each phone.
- Step 13** Configure the Busy Lamp Field feature, if required. You must use customized phone button templates to configure BLF/SpeedDial buttons.
- Step 14** Assign services to phone buttons, if required.
- Step 15** Provide power, install, verify network connectivity, and configure network settings for the Cisco Unified IP Phone.
- Step 16** Associate user with the phone (if required).
- Step 17** Make calls with the Cisco Unified IP Phone.
-

Supported Cisco Unified IP Phones

Table 36-3 provides an overview of the features that are available on the following Cisco Unified IP Phones that Cisco Unified Communications Manager supports:

- Cisco Unified IP Phone 6900 Series
- Cisco Unified IP Phone 7900 Series

- Cisco Unified IP Phone 8900 Series (SIP)
- Cisco Unified IP Phone 9900 Series (SIP)
- Cisco Unified IP Video Phone 7985 (SCCP)
- Cisco Unified IP Phone Expansion Module 7915 and 7916
- Cisco Unified IP Color Key Expansion Module
- Cisco IP Conference Station 7935, 7936, and 7937 (SCCP)
- Cisco Unified Wireless IP Phone 7921 and 7925 (SCCP)
- Cisco E20

For the latest information on features and services that these phone models support, see the following documentation:

- Phone administration or user documentation that supports the phone model and this version of Cisco Unified Communications Manager
- Firmware release notes for your phone model
- Cisco Unified Communications Manager release notes

**Note**

Phone models that are End of Software Maintenance will continue to be supported for the latest Unified Communications Manager releases, but they will not take advantage of any new Unified Communications Manager or firmware features associated with that release. For more information on End of Sale phone models, reference the model's End of Sale announcement for information on the level of firmware and hardware support.

Table 1: Supported Cisco Unified IP Phones and Features

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 9971 and 9951	<p>The Cisco Unified IP Phone 9971 and 9951 are advanced collaborative media endpoints that provide voice, video, applications, and accessories. Highlights include interactive multiparty video, high-resolution color touchscreen display, High-definition voice (HD voice), desktop Wi-Fi connectivity, Gigabit Ethernet and a new ergonomic design and user interface designed for simplicity and high usability. Accessories, which are sold separately, include the Cisco Unified Video Camera and the Cisco Unified IP Color Key Expansion Module.</p> <p>The Cisco Unified IP Phone 9971 supports the following buttons:</p> <ul style="list-style-type: none"> • Six feature buttons with state-indicating LEDs • Six call-session buttons with state-indicating LEDs • Applications, Directories, and Voicemail • Conference, Transfer, and Hold • Volume Up or Down • Back-lit Mute, speakerphone, and headset • Back, End Call, and 5-way navigation pad <p>The Cisco Unified IP Phone 9951 supports the following buttons:</p> <ul style="list-style-type: none"> • Five feature buttons with state-indicating LEDs • Five call-session buttons with state-indicating LEDs • Applications, Directories, and Voicemail • Conference, Transfer, and Hold • Volume Up or Down • Back-lit Mute, speakerphone, and headset • Back, End Call, and 5-way navigation pad <p>Both endpoints support Session Initiation Protocol (SIP).</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 8961	<p>The Cisco Unified IP Phone 8961 (SIP) is an advanced professional media endpoint that delivers an enhanced user experience with an easy-to-use and eco-friendly ergonomic design. Highlights of the portfolio include introduction of higher-resolution (VGA) color displays, a USB port, Gigabit Ethernet connectivity, and High-definition (HD) voice support, enabling a more productive user experience for multimedia application engagement. Application support includes XML and MIDlet-enabled applications. The Cisco Unified IP Phone 8961 is an ideal solution for knowledge professionals, administrative managers, and executives.</p> <p>The Cisco Unified IP Phone 8961 supports the following buttons:</p> <ul style="list-style-type: none"> • Five programmable feature buttons with state-indicating LEDs • Five call-session buttons with state-indicating LEDs • Applications, Directories, and Voicemail • Conference, Transfer, and Hold • Volume Up/Down, • Back-lit Mute, Speakerphone, and Headset • Back, End Call, and 5-Way Navigation Pad <p>Cisco Unified IP Phone 8961 supports Session Initiation Protocol (SIP).</p>
Cisco Unified IP Phone 8945	<p>The Cisco Unified IP Phone 8945 delivers affordable, business-grade voice and video communication services to customers worldwide.</p> <p>The Cisco Unified IP Phone 8945 has the following features:</p> <ul style="list-style-type: none"> • The phone delivers VGA presentation for calling, video calling, and applications, in addition to a 5-inch (10-cm) graphical TFT color display, 16-bit color depth, 640 x 480 effective pixel resolution, and backlighting. The display also supports localization requiring double-byte Unicode encoding for fonts • The phone supports four lines and four context-sensitive soft keys along with a high-definition voice, full-duplex speakerphone for a more productive and more flexible endpoint experience • Fixed keys for hold, transfer, redial, and conference; a tri-color LED line; and feature keys also make the endpoint simpler and easier to use. • The Cisco Unified IP Phone 8945 supports right-to-left language presentation on its display, addressing the language localization needs of global customers. <p>Cisco Unified IP Phone 8941 supports SCCP and SIP.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 8941	<p>The Cisco Unified IP Phone 8941 delivers affordable, business-grade voice and video communication services to customers worldwide.</p> <p>The Cisco Unified IP Phone 8941 has the following features:</p> <ul style="list-style-type: none"> • The phone delivers VGA presentation for calling, video calling, and applications, in addition to a 5-inch (10-cm) graphical TFT color display, 16-bit color depth, 640 x 480 effective pixel resolution, and backlighting. The display also supports localization requiring double-byte Unicode encoding for fonts • The phone supports four lines and four context-sensitive soft keys along with a high-definition voice, full-duplex speakerphone for a more productive and more flexible endpoint experience • Fixed keys for hold, transfer, redial, and conference; a tri-color LED line; and feature keys also make the endpoint simpler and easier to use. • The Cisco Unified IP Phone 8941 supports right-to-left language presentation on its display, addressing the language localization needs of global customers. <p>Cisco Unified IP Phone 8945 supports SCCP and SIP.</p>
Cisco Unified IP Phone 7975	<p>The Cisco Unified IP Phone 7975 demonstrates the latest advances in VoIP telephony, including wideband audio support, backlit color touchscreen display, and an integrated Gigabit Ethernet port.</p> <ul style="list-style-type: none"> • This IP phone includes a large, backlit, easy-to-read color display for easy access to communication information, timesaving applications, and features such as date and time, calling party name, calling party number, digits dialed, and presence information. • The phone provides direct access to eight telephone lines (or combination of lines, speed dials, and direct access to telephony features), five interactive softkeys that guide you through call features and functions, and an intuitive four-way (plus Select key) navigation cluster. • A hands-free speakerphone and handset designed for high-fidelity wideband audio are standard, as is a built-in headset connection. <p>Cisco Unified IP Phone 7975 supports SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7965	<p>The Cisco Unified IP Phone 7965 demonstrates the latest advances in VoIP telephony, including wideband audio support, backlit color display, and an integrated Gigabit Ethernet port.</p> <ul style="list-style-type: none"> • This IP phone includes a large, backlit, easy-to-read color display for easy access to communication information, timesaving applications, and features such as date and time, calling party name, calling party number, digits dialed, and presence information. • The phone provides direct access to six telephone lines (or combination of lines, speed dials, and direct access to telephony features), four interactive softkeys that guide you through call features and functions, and an intuitive four-way (plus Select key) navigation cluster. • A hands-free speakerphone and handset designed for high-fidelity wideband audio are standard, as is a built-in headset connection. <p>Cisco Unified IP Phone 7965 supports SCCP and SIP protocols.</p>
Cisco Unified IP Phone 7962	<p>The Cisco Unified IP Phone 7962 is a full-featured IP phone with speakerphone and handset designed for wideband audio. It is intended to meet the needs of managers and administrative assistants.</p> <ul style="list-style-type: none"> • It has six programmable backlit line/feature buttons and four interactive softkeys that guide you through all call features and functions. • The phone has a large, 4-bit grayscale graphical LCD that provides features such as date and time, calling party name, calling party number, digits dialed, and presence information. • The crisp graphic capability of the display allows for the inclusion of higher value, more visibly rich Extensible Markup Language (XML) applications, and support for localization requiring double-byte Unicode encoding for fonts. • A hands-free speakerphone and handset designed for hi-fidelity wideband audio are standard, as is a built-in headset connection and an integrated Ethernet switch. <p>Cisco Unified IP Phone 7962 supports SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7962	<p>The new Cisco Unified IP Phone 7941G-GE delivers the latest technology and advancements in Gigabit Ethernet IP telephony. This phone not only offers enhanced functionality for businesses that require advanced communications capabilities, but also brings network data and applications to users quickly with its Gigabit Ethernet port for integration to a PC or desktop server. The Cisco Unified IP Phone 7941G-GE is standards-based to deliver better interoperability and greater deployment flexibility.</p> <p>This state-of-the-art Gigabit Ethernet IP phone offers the same features as the Cisco Unified IP Phone 7941G and includes:</p> <ul style="list-style-type: none"> • High-resolution, graphical 4-bit grayscale display (320 x 222) that supports double-byte characters and Unicode text to benefit Extensible Markup Language (XML) application developers [Note: This phone requires IEEE 802.3af inline power or the use of a local power cube] • A full-featured handset that provides two programmable line and feature buttons • Four interactive softkeys to help guide users through various call features and functions
Cisco Unified IP Phone 7961G-GE	<p>The new Cisco Unified IP Phone 7961G-GE delivers the latest technology and advancements in Gigabit Ethernet IP telephony. This phone not only offers enhanced functionality for managers that require advanced communications capabilities, it also brings network data and applications to users quickly with its Gigabit Ethernet port for integration to a PC or desktop server. The Cisco Unified IP Phone 7961G-GE is standards-based to deliver better interoperability and greater deployment flexibility. This state-of-the-art Gigabit Ethernet IP phone also offers the same features as the Cisco Unified IP Phone 7961G, including:</p> <ul style="list-style-type: none"> • High-resolution, graphical 4-bit grayscale display (320 x 222) that supports double-byte characters and Unicode text to benefit Extensible Markup Language (XML) application developers [Note: This phone requires IEEE 802.3af inline power or the use of a local power cube] • A full-featured handset with six programmable line and feature buttons • Four interactive softkeys to help guide users through various call features and functions

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7960	<p>The Cisco Unified IP Phone 7960, a full-featured, six-line business set, supports SCCP and SIP and the following features:</p> <ul style="list-style-type: none"> • A help (?) button • Six programmable buttons to use as line, speed-dial, or feature buttons • Four fixed buttons for accessing voice-messaging messages, adjusting phone settings, accessing services, and accessing directories • Four softkeys for accessing additional call details and functionality (Softkeys change depending on the call state for a total of 16 softkeys.) • A large LCD display that shows call details and softkey functions • An internal, two-way, full-duplex speakerphone and microphone mute
Cisco Unified IP Phone 7945	<p>The Cisco Unified IP Phone 7945 demonstrates the latest advances in VoIP telephony, including wideband audio support, backlit color display, and an integrated Gigabit Ethernet port.</p> <p>This IP phone includes a large, backlit, easy-to-read color display for easy access to communication information, timesaving applications, and features such as date and time, calling party name, calling party number, digits dialed, and presence information.</p> <p>The phone provides direct access to two telephone lines (or combination of lines, speed dials, and direct access to telephony features), four interactive softkeys that guide you through call features and functions, and an intuitive four-way (plus Select key) navigation cluster.</p> <p>A hands-free speakerphone and handset designed for high-fidelity wideband audio are standard, as is a built-in headset connection.</p> <p>Cisco Unified IP Phone 7945 supports SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7942	<p>The Cisco Unified IP Phone 7942 is a full-featured IP phone with speakerphone and handset designed for wideband audio. It is intended to meet the needs of needs of transaction-type workers with significant phone traffic.</p> <p>It has two programmable backlit line/feature buttons and four interactive soft keys that guide you through all call features and functions.</p> <p>The phone has a large, 4-bit grayscale graphical LCD that provides features such as date and time, calling party name, calling party number, digits dialed, and presence information.</p> <p>The crisp graphic capability of the display allows for the inclusion of higher value, more visibly rich Extensible Markup Language (XML) applications, and support for localization requiring double-byte Unicode encoding for fonts.</p> <p>A hands-free speakerphone and handset designed for hi-fidelity wideband audio are standard, as is a built-in headset connection and an integrated Ethernet switch.</p> <p>Cisco Unified IP Phone 7942 supports SCCP and SIP protocols.</p>
Cisco Unified IP Phone 7941G	<p>The new Cisco Unified IP Phone 7941G-GE delivers the latest technology and advancements in Gigabit Ethernet IP telephony. This phone not only offers enhanced functionality for businesses that require advanced communications capabilities, but also brings network data and applications to users quickly with its Gigabit Ethernet port for integration to a PC or desktop server. The Cisco Unified IP Phone 7941G-GE is standards-based to deliver better interoperability and greater deployment flexibility.</p> <p>This state-of-the-art Gigabit Ethernet IP phone offers the same features as the Cisco Unified IP Phone 7941G and includes:</p> <ul style="list-style-type: none"> • High-resolution, graphical 4-bit grayscale display (320 x 222) that supports double-byte characters and Unicode text to benefit Extensible Markup Language (XML) application developers [Note: This phone requires IEEE 802.3af inline power or the use of a local power cube] • A full-featured handset that provides two programmable line and feature buttons • Four interactive softkeys to help guide users through various call features and functions

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7940	<p>The Cisco Unified IP Phone 7940, a two-line business set with features similar to the Cisco Unified IP Phone 7960, supports SCCP and SIP and includes the following features:</p> <ul style="list-style-type: none"> • A help (?) button • Two programmable buttons to use as line, speed-dial, or feature buttons • Four fixed buttons for accessing voice-messaging messages, services, and directories and for adjusting phone settings • Four softkeys for accessing additional call details and functionality (Softkeys change depending upon the call state for a total of 16 softkeys.) • A large LCD that shows call details and softkey functions • An internal, two-way, full-duplex speakerphone and microphone mute
Cisco Unified IP Phone 7931	<p>The Cisco Unified IP Phone 7931, designed for users who are familiar with traditional key sets, functions much like a digital business phone, allowing users to place and receive phone calls and to access features such as mute, hold, transfer, speed dial, call forward, and more, including</p> <ul style="list-style-type: none"> • Pixel-based backlit display • 24 configurable line buttons • Wideband Headset option-disabled by default (should be enabled only if the user headset supports wideband) • Abbreviated dialing • Audible Message Waiting Indicator • Call forward configurable display • Call forward destination override • Call Recording • Directed Call Park • Do Not Disturb (DND) • Video support • Voice Unified system

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7926G	<p>Increase the responsiveness of your mobile workforce within the campus and reduce overall costs with the Cisco Unified Wireless IP Phone 7926G. This portable, wireless IP phone delivers Cisco Unified Communications, integrated bar-code scanning, and custom applications.</p> <p>Building on the capabilities of the Cisco Unified Wireless IP Phone, the 7926G includes:</p> <ul style="list-style-type: none"> • Integrated EA11 2D bar-code scanner to track location, progress, and inventory • Ability to decode bar code symbologies; configure with Cisco Unified Communications Manager • Mobile Information Device Profile custom applications (MIDlets) for faster response • Specially designed holster and leather case
Cisco Unified Wireless IP Phone 7925G EX	<p>The Cisco Unified Wireless IP Phone 7925G-EX delivers all of the capabilities of the Cisco Unified Wireless IP Phone 7925G with the ruggedness and resiliency that is certified for deployment in potentially explosive environments such as chemical and manufacturing plants, utilities, and oil refineries.</p> <p>Features include:</p> <ul style="list-style-type: none"> • Atmospheres Explosibles (ATEX) Zone 2/Class 22 and Canadian Standards Association (CSA) Class I Division II certifications • IP64 rating for superior dust resistance with splashing water resistance adds resiliency. • Industry-standard yellow styling offers fast recognition in event of an emergency. • 802.11a/b/g standards for voice over WLAN (VoWLAN) communications support. • Supports third-party Bluetooth 2.0 headsets for added freedom • Large 2-inch color (176 x 220 pixel) display makes viewing easy. • Exceptional voice quality with high-definition voice (HD voice) • Built-in full-duplex speakerphone for high-quality hands-free communications. • Applications key provides direct access to XML applications such as push-to-talk and Lone Worker. • Extended-life batteries deliver a minimum of 13 hours talk time and up to 240 hours standby

Cisco Unified IP Phone Model	Description
Cisco Unified Wireless IP Phone 7925	<p>The Cisco Unified Wireless IP Phone 7925 is designed for users in rigorous workspaces as well as general office environments. It supports a wide range of features for enhanced voice communications, quality of service (QoS), and security. Some of the main benefits and highlights are listed here:</p> <ul style="list-style-type: none"> • IEEE 802.11 a/b/g radio • Two-inch color display • Bluetooth 2.0 support with Enhanced Data Rate (EDR) • IP54 rated for protection against dust and splashing water • MIL-STD-810F standard for shock resistance • Long battery life (up to 240 hours of standby time or 13 hours of talk time) • Built-in speakerphone for hands-free operation • Exceptional voice quality with support for wideband audio • Support for a wide range of applications through XML <p>Cisco Unified Wireless IP Phone 7925 supports the SCCP protocol.</p>
Cisco Unified Wireless IP Phone 7921	<p>The Cisco Unified Wireless IP Phone 7921 supports a host of calling features and voice-quality enhancements. The device is an advanced media IP phone, delivering wideband audio capabilities.</p> <p>In addition to wideband audio, Cisco Unified Wireless IP Phone 7921 supports presence, which enables users in a mobile Wi-Fi environment to view the current status of other users. Because the Cisco Unified Wireless IP Phone 7921G is designed to grow with system capabilities, features will keep pace with new system enhancements.</p> <p>Cisco Unified Wireless IP Phone 7921 supports the SCCP protocol.</p>
Cisco Unified Wireless IP Phone 7920	<p>The Cisco Wireless IP Phone 7920, which is an easy-to-use IEEE 802.11b wireless IP phone, provides comprehensive voice communication in conjunction with Cisco Unified Communications Manager and Cisco Aironet 1200, 1100, 350, and 340 series of Wi-Fi (IEEE 802.11b) access points. Features include</p> <ul style="list-style-type: none"> • A pixel-based display for intuitive access to calling features • Two softkeys that dynamically present calling options to the user • A four-way rocker switch that allows easy movement through the displayed information • Volume control for easy decibel-level adjustments of the handset and ringer when in use

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone Expansion Module 7914	<p>Cisco Unified IP Phone Expansion Module 7914 extend the functionality of a Cisco Unified IP Phone by providing 14 additional buttons. To configure these buttons as line or speed dials, use Phone Button Template Configuration.</p> <p>Note You can create the Cisco Unified IP Phone Expansion Module 7914 phone button template by copying the phone button template for the standard Cisco Unified IP Phone phone model that you are using with your Cisco Unified IP Phone Expansion Module 7914.</p> <p>The Cisco Unified IP Phone Expansion Module 7914 includes an LCD to identify the function of the button and the line status.</p> <p>You can daisy chain two Cisco Unified IP Phone Expansion Modules 7914 to provide 28 additional lines or speed-dial and feature buttons.</p>
Cisco Unified IP Phone Expansion Module 7915 and Cisco Unified IP Phone Expansion Module 7916	<p>Cisco Unified IP Phone Expansion Module 7915 and 7916 extends the functionality of a Cisco Unified IP Phone by providing 24 additional buttons. To configure these buttons as line or speed dials, use Phone Button Template Configuration.</p> <p>Note You create the Cisco Unified IP Phone Expansion Module phone button template by copying the phone button template for the standard Cisco Unified IP Phone phone model that you are using with your Cisco Unified IP Phone Expansion Module 7915 or 7916.</p> <p>The Cisco Unified IP Phone Expansion Module 7915 and 7916 includes an LCD to identify the function of the button and the line status.</p> <p>You can daisy chain two Cisco Unified IP Phone Expansion Module 7915s or 7916s to provide 48 additional lines or speed-dial and feature buttons.</p>
Cisco Unified IP Color Key Expansion Module	<p>Cisco Unified IP Color Key Expansion Module extends the functionality of a Cisco Unified IP Phone by providing 36 additional buttons. The programmable buttons can be set up as phone line buttons, speed-dial buttons, or phone feature buttons. To configure these buttons as line buttons, speed dial buttons, or phone features buttons, use the Phone Button Template Configuration.</p> <p>Note You create the Cisco Unified IP Color Key Expansion Module phone button template by copying the phone button template for the standard Cisco Unified IP Phone model that you are using with your Cisco Unified IP Color Key Expansion Module.</p> <p>You can attach one Cisco Unified IP Color Key Expansion Module to a Cisco Unified IP Phone 8961 for 36 additional buttons, two Cisco Unified IP Color Key Expansion Modules to a Cisco Unified IP Phone 9951 for 72 additional buttons, and three Cisco Unified IP Color Key Expansion Modules to a Cisco Unified IP Phone 9971 for 108 additional buttons.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7911	<p>The Cisco Unified IP Phone 7911, which is a single-line phone that supports a maximum of six calls at the same time, supports SCCP and SIP and provides basic-feature functionality for individuals who conduct low to medium telephone traffic.</p> <p>The Applications Menu button opens up a main applications menu.</p> <p>This phone, which supports inline power, provides an integrated 10/100 Ethernet switch for connectivity to a collocated PC.</p> <p>This phone offers four dynamic softkeys.</p>
Cisco Unified IP Phone 7906	<p>The Cisco Unified IP Phone 7906, which is a single-line phone that supports a maximum of six calls at the same time, supports SCCP and SIP and provides basic-feature functionality for individuals who conduct low to medium telephone traffic.</p> <p>The Applications Menu button opens up a main applications menu.</p> <p>This phone, which supports inline power, provides an integrated 10/100 Ethernet switch for connectivity to a collocated PC.</p> <p>This phone offers four dynamic softkeys.</p>
Cisco Unified IP Phone 7985	<p>The Cisco Unified IP Phone 7985G provides business-quality video over the same data network that your computer uses. The video phone provides the same softkey functionality and features as a Cisco Unified IP Phone, which allows you to place and receive calls, put calls on hold, transfer calls, make conference calls, and so on. The Cisco Unified IP Phone 7985G provides the following features:</p> <ul style="list-style-type: none"> • Color screen • Support for up to eight line or speed-dial numbers • Context-sensitive online help for buttons and feature

Cisco Unified IP Phone Model	Description
Cisco Unified IP Conference Station 7937	<p>The Cisco Unified IP Conference Station 7937 combines state-of-the-art wideband speakerphone conferencing technologies with award-winning Cisco voice communication technologies. The net result is a conference room phone that offers superior wideband voice and microphone quality, with simplified wiring and administrative cost benefits.</p> <p>A full-featured, IP-based, hands-free conference station, the Cisco Unified IP Conference Station 7937 is designed for use on desktops, in conference rooms, and in executive suites.</p> <p>Cisco Unified IP Conference Station 7937 features include:</p> <ul style="list-style-type: none"> • Superior wideband acoustics with the support of the G.722 wideband codec • Support for IEEE Power over Ethernet (PoE) or the Cisco Power Cube 3 • Expanded room coverage up to 30 feet by 40 feet with the optional external microphone kit • Support for a third-party lapel microphone kit1 • New larger backlit liquid crystal display (LCD) • Global localization <p>Cisco Unified IP Conference Station 7937 supports the SCCP protocol.</p>
Cisco Unified IP Conference Station 7936	<p>The Cisco Unified IP Conference Station 7936, a full-featured, IP-based, hands-free conference station for use on desktops, in offices, and in small- to medium-sized conference rooms, includes the following features:</p> <ul style="list-style-type: none"> • Three softkeys and menu navigation keys that guide a user through call features and functions including available features Call Park, Call Pickup, Group Call Pickup, Transfer, and Conference (Ad Hoc and Meet-Me). • An LCD that indicates the date and time, calling party name, calling party number, digits dialed, feature, and line status • A digitally tuned speaker and three microphones that allow conference participants to move around while speaking • Microphone mute • Ability to add external microphones to support larger rooms

Cisco Unified IP Phone Model	Description
Cisco IP Conference Station 7935	<p>The Cisco IP Conference Station 7935, a full-featured, IP-based, hands-free conference station for use on desktops, in offices, and in small- to medium-sized conference rooms, includes the following features:</p> <ul style="list-style-type: none"> • Three softkeys and menu navigation keys that guide a user through call features and functions Available features include Call Park, Call Pickup, Group Call Pickup, Transfer, and Conference (Ad Hoc and Meet-Me). • An LCD that indicates the date and time, calling party name, calling party number, digits dialed, feature, and line status • A digitally tuned speaker and three microphones that allow conference participants to move around while speaking • Microphone mute
Cisco Unified IP Phone 6961	<p>The Cisco Unified IP Phone 6961 is a new and innovative IP endpoint that delivers affordable, business-grade voice communication and video communication services to customers worldwide.</p> <ul style="list-style-type: none"> • The Cisco Unified IP Phone 6961 supports 12 lines, paper label inserts for line and feature descriptions along with a full-duplex speakerphone for a more productive, more flexible, and easier-to-use endpoint experience. • Single-call per-line appearance is introduced, delivering traditional telephony-like user experience for customers who seek this type of call interaction for their users. • Fixed keys for hold, transfer, and conference; tri-color LED line and feature keys also make the endpoint simpler and easier to use • Right-to-left language presentation is also supported on the displays, addressing the language localization needs of global customers. • The Cisco Unified IP Phone 6961 is also energy-efficient and eco-friendly, in support of customer green initiatives. A Deep-Sleep option provides energy savings. With this option, the Cisco Unified IP Phone 6961 consumes up to 50 percent less power in off-hours versus when the phone is idle during normal business hours. In addition, the Cisco Unified IP Phone 6961 employs use of both recyclable and reground plastics for a more earth-responsible solution. <p>Cisco Unified IP Phone 6961 supports the SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 6941	<p>The Cisco Unified IP Phone 6941 is an innovative IP endpoint that delivers affordable, business-grade voice communication and support for video communications services to customers worldwide.</p> <ul style="list-style-type: none"> • The Cisco Unified IP Phone 6941 supports four lines and a full-duplex speakerphone for a more productive, more flexible, and easier-to-use endpoint experience. • The phone supports single-call per-line appearance, offering traditional telephony-like user experience for customers who seek this type of call interaction for their users. • Fixed keys for hold, transfer, and conference; tri-color LED line and feature keys also make the phone simpler and easier to use. • Right-to-left language presentation is also supported on the displays, addressing the language localization needs of global customers. • The Cisco Unified IP Phone 6941 is also energy-efficient and eco-friendly, in support of customer green initiatives. A Deep-Sleep option provides energy savings. With this option, the phone consumes up to 50 percent less power in off-hours versus when the phone is idle during normal business hours. In addition, reground and recyclable plastics deliver a more earth-responsible solution. <p>Cisco Unified IP Phone 6941 supports the SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 6921	<p>The Cisco Unified IP Phone 6921 is an innovative endpoint that delivers affordable, business-grade voice communications and support for video communications services to customers worldwide.</p> <ul style="list-style-type: none">• The Cisco Unified IP Phone 6921 supports two lines and offers a full-duplex speakerphone for a more productive, more flexible, and easier-to-use endpoint experience.• The phone supports single-call per-line appearance, offering traditional telephony-like user experience for customers who seek this type of call interaction for their users.• Fixed keys for hold, transfer, and conference; tri-color LED line and feature keys also make the phone simpler and easier to use.• Right-to-left language presentation is also supported on the displays, addressing the language localization needs of global customers.• The Cisco Unified IP Phone 6921 is also energy-efficient and eco-friendly, in support of customer green initiatives. A Deep-Sleep option provides energy savings. With this option, the phone consumes up to 50 percent less power in off-hours versus when the phone is idle during normal business hours. In addition, reground and recyclable plastics deliver a more earth-responsible solution. <p>Cisco Unified IP Phone 6921 supports the SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 6911	<p>The Cisco Unified IP Phone 6911 is a single-line endpoint delivering affordable access to Cisco voice communication services. It is an ideal solution for light communication requirements. Examples include classrooms, manufacturing floors, or employees in cubicles or teleworking from home.</p> <ul style="list-style-type: none"> • The Cisco Unified IP Phone 6911 supports two incoming calls with a single-line endpoint. • A full-duplex speakerphone is included in the design, which provides a more productive, flexible, and easier-to-use endpoint experience. • Integrated IEEE 10/100 Ethernet switch ports support connection to a co-located PC while reducing cabling infrastructure and administration costs. • The phone includes fixed keys for hold, transfer, conference, redial, and voicemail, making the phone simple and easy-to-use. In addition, a programmable feature key is supported for quick access to advanced communication services. • Tri-color LED illuminates on the line key to provide quick call-state indication at a glance. • The Cisco Unified IP Phone 6911 is also eco-friendly, taking advantage of reground and recyclable plastics to deliver a more earth-responsible solution. <p>Cisco Unified IP Phone 6911 supports the SCCP and SIP protocols.</p>
Cisco Unified IP Phone 6901	<p>The Cisco Unified IP Phone 6901 is a single-line endpoint delivering cost-effective access to Cisco Unified Communications. Designed with a trimline-like low profile, the phone is an ideal solution for lobbies, hallways, elevators, hotel bathrooms, or other settings that have an occasional need for voice communications services.</p> <ul style="list-style-type: none"> • The phone supports two incoming calls with call-waiting service. • Fixed feature keys provide one-touch access to Hold, Redial, and Call Waiting. • Transfer and Conference can be supported by using the hook-switch similar to that of traditional analog phones. • The Cisco Unified IP Phone 6901 is an earth-friendly solution. As with the other Cisco Unified IP Phone 6900 Series endpoints, the Cisco Unified IP Phone 6901 takes advantage of reground and recyclable plastics for a more earth-responsible solution. <p>Cisco Unified IP Phone 6901 supports the SCCP and SIP protocols.</p>

Cisco Unified IP Phone Model	Description
Cisco Unified SIP Phone 3951	<p>Be aware that the Cisco Unified SIP Phone 3951, a low-end phone that runs SIP, is available only in Asia Pacific and Latin American countries. For more information, contact your Cisco representative.</p>
Cisco Unified SIP Phone 3905	<p>Replace your existing analog and digital phone deployments with affordable IP communication endpoints using the Cisco Unified SIP Phone 3905.</p> <p>The Cisco Unified SIP Phone 3905 includes interactive features such as:</p> <ul style="list-style-type: none"> • Single-line IP phone with support for up to two concurrent calls • Graphical 128x32-pixel monochrome display with a two-way navigation button • Full duplex speakerphone for flexibility with hands-free communications • Fixed keys for common telephony features: hold, redial, transfer, and mute • Foldable, single-position display stand to simplify wall-mount deployments
Cisco Unified SIP Phone 3911	<p>The Cisco Unified SIP Phone 3911 is a cost-effective, entry-level phone that addresses the needs of a lobby, laboratory, manufacturing floor, or hallway. This single-line phone with speakerphone and internal microphone can also fill the communication needs of cubicle, retail, classroom, or manufacturing workers or anyone with low to moderate telephone needs.</p> <p>The Cisco Unified SIP Phone 3911 provides:</p> <ul style="list-style-type: none"> • Fixed feature keys that provide one-touch access to redial, transfer, conference, hold, line select, mute, speakerphone, and voicemail access features • A display that supports additional capabilities such as caller ID, call history, and phone configuration • Choice of IEEE 802.3af Power over Ethernet (PoE) or local power through an optional power adaptor

Cisco Unified IP Phone Model	Description
Cisco Unified SIP Phone 3905	<p>Replace your existing analog and digital phone deployments with affordable IP communication endpoints using the Cisco Unified SIP Phone 3905.</p> <p>This phone also gives you access to the comprehensive suite of capabilities supported by Cisco Unified Communications Manager.</p> <p>The Cisco Unified SIP Phone 3905 includes interactive features such as:</p> <ul style="list-style-type: none">• Single-line IP phone with support for up to two concurrent calls• Graphical 128x32-pixel monochrome display with a two-way navigation button• Full duplex speakerphone for flexibility with hands-free communications• Fixed keys for common telephony features: hold, redial, transfer, and mute• Foldable, single-position display stand to simplify wall-mount deployments

Cisco Unified IP Phone Model	Description
Cisco E20	<p>The Cisco E20 reinvents the desk phone by merging voice, video, and collaboration into one device. A highly scalable solution for enterprise mass deployment, users will immediately see the benefits of increased productivity and daily collaboration.</p> <p>The Cisco E20 offers the following capabilities:</p> <ul style="list-style-type: none"> • Intuitive user interface and keypad for quick access to all IP phone and video services • Familiar telephony features such as Hold, Transfer, Resume, and Conference • Handset, headset (bluetooth), speakerphone flexibility • Navigation cluster with select button • Message waiting indicator/button • 5 contextual softkeys • USB picture frame • High-resolution camera with integrated privacy shutter • DVD quality, w448p video resolution • 10.6" wide format LCD display with WXGA resolution • Audio standards: MPEG4 AAC-LD, G.729ab, G.722, G.722.1, G.711 • Video standards and resolutions: H.264, H.263, and H.263+ from SIF up to w448p • Bandwidth up to 1152 kbps <p>The Cisco E20 supports SIP.</p>

Third-Party SIP Endpoints

Cisco Unified Communications Manager supports a variety of third-party SIP endpoints, which are configured in Cisco Unified Communications Manager Administration, Phone Configuration.

Cisco Unified Communications Manager requires user licenses. These licenses get configured in Cisco Unified Communications Manager Administration, License Configuration. When acquiring user licenses, the administrator purchases one user license, called user connect license (UCL), for each user. License Configuration uses device license units (DLU). For example, if there are three generic desktop video endpoint users (3 UCLs), License Configuration would need 18 DLUs (3 UCL x 6 DLU = 18 DLU).

When using Phone Configuration to add a third-party SIP endpoint, the following device phone types are available:

- Third-Party SIP Device (Advanced)-This eight-line SIP device is an RFC3261-compliant phone that is running SIP from third-party companies; this device requires 6 DLUs.
- Third-Party SIP Device (Basic)-This one-line SIP device is an RFC3261-compliant phone that is running SIP from third-party companies; this device requires 3 Device License Units (DLUs).
- Third-Party AS-SIP Device - Third-party AS-SIP endpoints are compliant with Assured Services SIP, which includes MLPP, DSCP, TLS/SRTP, and IPv6 requirements.
- Generic Desktop Video Endpoint-This SIP device supports video, security, configurable trust, and Cisco extensions; this device requires 6 DLUs. This device supports 8 lines; the maximum number of calls and busy trigger for each line is 4 and 2, respectively.
- Generic Single Screen Room System-This SIP device supports single screen telepresence (room systems), video, security, configurable trust, and Cisco extensions; this device requires 6 DLUs. This device supports 8 lines; the maximum number of calls and busy trigger for each line is 4 and 2, respectively.
- Generic Multiple Screen Room System-This SIP device supports multiple screen telepresence (room systems), video, security, configurable trust, and Cisco extensions; this device requires 6 DLUs. This device supports 8 lines; the maximum number of calls and busy trigger for each line is 4 and 2, respectively.

H.323 Clients and CTI Ports

Cisco Unified Communications Manager Administration enables you to configure software-based devices such as H.323 clients and CTI ports. Software-based Cisco Unified Communications Manager applications such as Cisco IP Softphone, Cisco Unified Communications Manager Auto-Attendant, and Cisco IP Interactive Voice Response (IVR) use CTI ports that are virtual devices.

H.323 clients include Microsoft NetMeeting devices.

You configure H.323 clients and CTI ports through the Phone Configuration window in Cisco Unified Communications Manager Administration like you do phones, but they often require fewer configuration settings.



Note

Cisco recommends that you do not configure CTI ports or devices that use TAPI applications in a line group.

For information on H.323 clients and shared line appearances, see the [Shared Line Appearance](#).

CTI Remote Device Setup

The CTI Remote Device type enables third-party desktop clients to receive incoming calls, initiate Dial via Office reverse calls, and perform mid-call features. Consult the third-party vendor documentation to confirm support for this device type.

In Cisco Unified Communications Manager Administration, use the **Device > Phone** menu path to configure CTI Remote Device. CTI Remote devices configuration specifies a set of parameters that apply to all the CTI Remote Devices for the user.

CTI Remote Device type represents the users remote device(s), similar to the Mobile Communicator device type. You can add a Remote Destination for a CTI Remote Device. The Remote Destination associated with the CTI Remote Device specifies the number to reach the Remote Device. The maximum number of Remote Destinations that you can configure for a CTI Remote Device is dependent on the Remote Destination limit set for the Owner User ID. By default, this value is set to 4.

Tips About Configuring CTI Remote Devices

You can add a maximum of five Directory Numbers to the CTI Remote Device. To register a CTI Remote Device, add a Directory Number to that device. You cannot register a CTI Remote Device without a Directory Number.

Using the GUI

For instructions on how to use the Cisco Unified Communications Manager Administration Graphical User Interface (GUI) to find, delete, configure, or copy records, see the *Cisco Unified Communications Manager Administration Guide* and its subsections, which explain how to use the GUI and detail the functions of the buttons and icons.

Configuration Settings Table

The following table describes the available settings to configure a CTI remote device through the Phone Configuration Settings window.

Table 2: CTI Remote Device Settings

Field	Description
CTI Remote Device Information	
Device Information	
Registration	Specifies the registration status of the CTI Remote Device.
Device Status	Specifies if the device is active or inactive.
Device Trust	Specifies if the device is trusted.
Active Remote Destination	Specifies the Remote Destination which is active. The CTI client can specific one remote destination as 'active' at any one given time. Incoming calls and Dial via Office (DVO) calls are routed to the active remote destination.
Owner User ID	From the drop-down list box, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device.

Field	Description
Device Name	<p>Specifies the name for the CTI Remote Device which is automatically populated based on the Owner User ID.</p> <p>The format of the device name is <i>CTIRD<OwnerUserID></i> by default.</p> <p>You can also edit the device name. The device name can comprise up to 15 characters. Valid characters include letters, numbers, dashes, dots (periods), spaces, and underscores.</p>
Description	<p>Enter a text description of the CTI remote device.</p> <p>This field can comprise up to 128 characters. You can use all characters except quotes ("), close angle bracket (>), open angle bracket (<), backslash (\), ampersand (&), and percent sign (%).</p>
Device Pool	<p>Select the device pool which defines the common characteristics for CTI remote devices.</p> <p>For more information on how to configure the device pool, see Device Pool Configuration Settings.</p>
Calling Search Space	<p>Using the drop-down list box, choose the calling search space or leave the calling search space as the default of <None>.</p>
User Hold MOH Audio Source	<p>Using the drop-down list box, choose the audio source to use for music on hold (MOH) when a user initiates a hold action.</p>
Network Hold MOH Audio Source	<p>Using the drop-down list box, choose the audio source to use for MOH when the network initiates a hold action.</p>
Location	<p>Using the drop-down list box, choose the location that is associated with the phones and gateways in the device pool.</p>
Calling Party Transformation CSS	<p>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device pool.</p>

Field	Description
Ignore Presentation Indicators (internal calls only)	Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Cisco Unified CM ignores any presentation restriction that is received for internal calls.
Call Routing Information	
Inbound/Outbound Calls Information	
Calling Party Transformation CSS	This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.
Use Device Pool Calling Party Transformation CSS	To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the Trunk Configuration window.
Protocol Specific Information	
Presence Group	<p>Configure this field with the Presence feature.</p> <p>If you are not using this application user with presence, leave the default (None) setting for presence group.</p> <p>From the drop-down list box, choose a Presence group for the application user. The group selected specifies the destinations that the application user, such as IPMASysUser, can monitor.</p>

Field	Description
SUBSCRIBE Calling Search Space	<p>Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user.</p> <p>From the drop-down list box, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list box.</p> <p>If you do not select a different calling search space for the end user from the drop-down list, the SUBSCRIBE calling search space defaults to None.</p> <p>To configure a SUBSCRIBE calling search space specifically for this purpose, you configure a calling search space as you do all calling search spaces.</p>
Rerouting Calling Search Space	<p>From the drop-down list box, choose a calling search space to use for rerouting.</p> <p>The rerouting calling search space of the referrer gets used to find the route to the refer-to target. When the Refer fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message.</p> <p>The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target.</p>
Do Not Disturb Information	
Do Not Disturb	Check this check box to enable Do Not Disturb on the remote device.
DND Option	When you enable DND on the phone, Call Reject option specifies that no incoming call information gets presented to the user. Depending on how you configure the DND Incoming Call Alert parameter, the phone may play a beep or display a flash notification of the call.

After you configure the CTI Remote Device, you can configure the associated remote destination. Click **Device > Phone > CTI Remote Device > Associated Remote Destinations > Add a New Remote Destination** to add and associate the remote destination with the CTI Remote Device.

**Note**

You can configure a maximum of four unique Remote Destinations to associate with the CTI Remote Device.

When the Remote Destination is configured through the CTI Remote Device configuration window, the following parameters are altered.

- Enable Move to Mobile—This function is disabled by default. The field cannot be edited and is not visible on the Administrative Interface.
- Enable Single Number Reach—This function is enabled by default. The field cannot be edited and is not visible on the Administrative Interface.

**Note**

This feature requires a Cisco Jabber client and this functionality is intended to be supported in Jabber for Windows 9.1.

You can also configure the remote destination from **Device > Remote Destination** window.

**Note**

You cannot edit these two fields while you configure the Remote Destination through the CTI Remote Device configuration window.

Client Services Framework Setup

In Cisco Unified Communications Manager Administration, use the **Device > Phone** menu path to configure the Cisco Unified Client Services Framework device.

**Note**

This section describes how to configure a Cisco Unified Client Services Framework device through the Phone Configuration Settings window.

For instructions on how to use the Cisco Unified Communications Manager Administration Graphical User Interface (GUI) to find, delete, configure, or copy records, see the *Cisco Unified Communications Manager Administration Guide* and its subsections, which explain how to use the GUI and detail the functions of the buttons and icons.

Configuration Settings Table

The following table describes the available settings in the Client Services Framework Configuration window.

Table 3: Client Services Framework Configuration Settings

Field	Description
Cisco Unified Client Services Framework Information	

Field	Description
Device Protocol	Specifies the protocol used to the Cisco Unified Client Services Framework.
Active Remote Destination	Specifies the Remote Destination which is active. The CSF client can specific one remote destination as 'active' at any one given time. Incoming calls and Dial via Office (DVO) calls are routed to the active remote destination.
Device Information	
Device Status	Specifies if the device is active or inactive.
Device Trust	Specifies if the device is trusted or not.
Device Name	Enter a text name for the Client Services Framework. This name can comprise up to 50 characters. Valid characters include letters, numbers, dashes, dots (periods), spaces, and underscores.
Description	Enter a text description of the Client Services Framework. This field can comprise up to 128 characters. You can use all characters except quotes ("), close angle bracket (>), open angle bracket (<), backslash (\), ampersand (&), and percent sign (%).
Device Pool	Select the device pool which defines the common characteristics for Client Services Framework. For more information on how to configure the device pool, see Device Pool Configuration Settings.
Common Device Configuration	Using the drop-down list box, choose the common device configuration to which you want this trunk assigned. The common device configuration includes the attributes (services or features) that are associated with a particular user. Common device configurations are configured in the Common Device Configuration window.
Phone Button Template	Using the drop-down list box, choose the appropriate phone button template. The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.
Common Phone Profile	Using the drop-down list box, choose the common phone profile to specify the data that is required by the Cisco TFTP.

Field	Description
Calling Search Space	<p>Choose the calling search space to be used for routing Mobile Voice Access or Enterprise Feature Access calls.</p> <p>Note This calling search space setting applies only when you are routing calls from the remote destination, which specifies the outbound call leg to the dialed number for Mobile Voice Access and Enterprise Feature Access calls.</p>
AAR Calling Search Space	<p>Choose the appropriate calling search space for the device to use when automated alternate routing (AAR) is performed. The AAR calling search space specifies the collection of route partitions that are searched to determine how to route a collected (originating) number that is otherwise blocked due to insufficient bandwidth.</p>
Media Resource Group List	<p>Choose the appropriate Media Resource Group List. A Media Resource Group List comprises a prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music On Hold server, from the available media resources according to the priority order that is defined in a Media Resource Group List.</p> <p>If you choose <none>, Cisco Unified Communications Manager uses the Media Resource Group that is defined in the device pool.</p>
User Hold MOH Audio Source	<p>Using the drop-down list box, choose the audio source to use for music on hold (MOH) when a user initiates a hold action.</p>
Network Hold MOH Audio Source	<p>Using the drop-down list box, choose the audio source to use for MOH when the network initiates a hold action.</p>
Location	<p>Using the drop-down list box, choose the location that is associated with the phones and gateways in the device pool.</p>
AAR Group	<p>Choose the automated alternate routing (AAR) group for this device. The AAR group provides the prefix digits that are used to route calls that are otherwise blocked due to insufficient bandwidth. An AAR group setting of None specifies that no rerouting of blocked calls will be attempted.</p>

Field	Description
User Locale	<p>From the drop-down list box, choose the locale that is associated with the CTI route point. The user locale identifies a set of detailed information to support users, including language and font.</p> <p>Cisco Unified Communications Manager makes this field available only for CTI route points that support localization.</p> <p>Note If no user locale is specified, Cisco Unified Communications Manager uses the user locale that is associated with the device pool.</p> <p>Note If the users require that information be displayed (on the phone) in any language other than English, verify that the locale installer is installed before configuring user locale. See the Cisco Unified Communications Manager locale installer that is in the Cisco Unified Communications Operating System Administration Guide.</p>
Network Locale	<p>From the drop-down list box, choose the locale that is associated with the gateway. The network locale identifies a set of detailed information to support the hardware in a specific location. The network locale contains a definition of the tones and cadences that the device uses in a specific geographic area.</p> <p>Note Choose only a network locale that is already installed and that the associated devices support. The list contains all available network locales for this setting, but not all are necessarily installed. If the device is associated with a network locale that it does not support in the firmware, the device will fail to come up.</p>

Field	Description
Device Mobility Mode	<p>From the drop-down list box, turn the device mobility feature on or off for this device or choose Default to use the default device mobility mode. Default setting uses the value for the Device Mobility Mode service parameter for the device.</p> <p>Click View Current Device Mobility Settings to display the current values of these device mobility parameters:</p> <ul style="list-style-type: none"> • Cisco Unified Communications Manager Group • Roaming Device Pool • Location • Region • Network Locale • AAR Group • AAR Calling Search Space • Device Calling Search Space • Media Resource Group List • SRST <p>For more configuration information, see “Device Mobility” in the <i>Cisco Unified Communications Manager Features and Services Guide</i>.</p>
Owner User ID	<p>From the drop-down list box, choose the user ID of the assigned phone user. The user ID gets recorded in the call detail record (CDR) for all calls made from this device.</p> <p>Note Do not configure this field if you are using extension mobility. Extension mobility does not support device owners.</p>
Mobility User ID	<p>From the drop-down list box, choose the user ID of the person to whom this dual-mode phone is assigned.</p> <p>Note The Mobility User ID configuration gets used for the Cisco Unified Mobility and Mobile Voice Access features for dual-mode phones.</p> <p>Note The Owner User ID and Mobility User ID can differ.</p>
Primary Phone	<p>Choose the physical phone that will be associated with the application, such as IP communicator or Cisco Unified Personal Communicator. When you choose a primary phone, the application consumes fewer device license units and is considered an “adjunct” license (to the primary phone). See “Licensing” in the Cisco Unified Communications Manager Features and Services Guide.</p>

Field	Description
Use Trusted Relay Point	<p>From the drop-down list box, enable or disable whether Cisco Unified CM inserts a trusted relay point (TRP) device with this media endpoint. Choose one of the following values:</p> <ul style="list-style-type: none"> • Default—If you choose this value, the device uses the Use Trusted Relay Point setting from the common device configuration with which this device associates. • Off—Choose this value to disable the use of a TRP with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates. • On—Choose this value to enable the use of a TRP with this device. This setting overrides the Use Trusted Relay Point setting in the common device configuration with which this device associates. <p>A Trusted Relay Point (TRP) device designates an MTP or transcoder device that is labeled as Trusted Relay Point. Cisco Unified CM places the TRP closest to the associated endpoint device if more than one resource is needed for the endpoint (for example, a transcoder or RSVPAgent).</p> <p>If both TRP and MTP are required for the endpoint, TRP gets used as the required MTP. See the “TRP Insertion” in <i>Cisco Unified Communications Manager</i> in the <i>Cisco Unified Communications Manager System Guide</i> for details of call behavior.</p> <p>If both TRP and RSVPAgent are needed for the endpoint, Cisco Unified CM first tries to find an RSVPAgent that can also be used as a TRP.</p> <p>If both TRP and transcoder are needed for the endpoint, Cisco Unified CM first tries to find a transcoder that is also designated as a TRP.</p> <p>See the “Trusted Relay Point” section and its subtopics in the “Media Resource Management” chapter of the <i>Cisco Unified Communications Manager System Guide</i> for a complete discussion of network virtualization and trusted relay points.</p>

Field	Description
Always Use Prime Line	<p>From the drop-down list box, choose one of the following options:</p> <ul style="list-style-type: none"> • Off—When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received. • On—When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls. • Default—Cisco Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter, which supports the Cisco Call Manager service.
Always Use Prime Line for Voice Message	<p>From the drop-down list box, choose one of the following options:</p> <ul style="list-style-type: none"> • On—If the phone is idle, the primary line on the phone becomes the active line for retrieving voice messages when the phone user presses the Messages button on the phone. • Off—If the phone is idle, pressing the Messages button on the phone automatically dials the voice-messaging system from the line that has a voice message. Cisco Unified CM always selects the first line that has a voice message. If no line has a voice message, the primary line gets used when the phone user presses the Messages button. • Default—Cisco Unified CM uses the configuration from the Always Use Prime Line for Voice Message service parameter, which supports the Cisco Call Manager service.
Calling Party Transformation CSS	<p>This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.</p> <p>Tip Before the call occurs, the device must apply the transformation by using digit analysis. If you configure the Calling Party Transformation CSS as None, the transformation does not match and does not get applied. Ensure that you configure the Calling Party Transformation Pattern in a non-null partition that is not used for routing.</p>
Geolocation	<p>From the drop-down list box, choose a geolocation.</p> <p>You can choose the Unspecified geolocation, which designates that this device does not associate with a geolocation.</p> <p>You can also choose a geolocation that has been configured with the System > Geolocation Configuration menu option.</p>

Field	Description
Ignore Presentation Indicators (internal calls only)	<p>Check this check box to configure call display restrictions on a call-by-call basis. When this check box is checked, Cisco Unified CM ignores any presentation restriction that is received for internal calls.</p> <p>Use this configuration in combination with the calling line ID presentation and connected line ID presentation configuration at the translation pattern level. Together, these settings allow you to configure call display restrictions to selectively present or block calling and/or connected line display information for each call.</p>
Allow Control of Device from CTI Control of Device from CTI	<p>Check this check box to allow CTI to control and monitor this device.</p> <p>If the associated directory number specifies a shared line, the check box should be enabled as long as at least one associated device specifies a combination of device type and protocol that CTI supports.</p>
Logged Into Hunt Group	<p>This check box, which gets checked by default for all phones, indicates that the phone is currently logged in to a hunt list (group). When the phone gets added to a hunt list, the administrator can log the user in or out by checking (and unchecking) this check box.</p> <p>Users use the softkey on the phone to log their phone in or out of the hunt list.</p>

Field	Description
Remote Device	<p>If you are experiencing delayed connect times over SCCP pipes to remote sites, check the Remote Device check box in the Phone Configuration window. Checking this check box tells Cisco Unified CM to allocate a buffer for the phone device when it registers and to bundle SCCP messages to the phone.</p> <p>Tip Because this feature consumes resources, be sure to check this check box only when you are experiencing signaling delays for phones that are running SCCP. Most users do not require this option.</p> <p>Cisco Unified CM sends the bundled messages to the phone when the station buffer is full, as soon as it receives a media-related message, or when the Bundle Outbound SCCP Messages timer expires.</p> <p>To specify a setting other than the default setting (100 msec) for the Bundle Outbound SCCP Messages timer, configure a new value in the Service Parameters Configuration window for the Cisco CallManager service. Although 100 msec specifies the recommended setting, you may enter 15 msec to 500 msec.</p> <p>The phone must support SCCP version 9 to use this option. The following phones do not support SCCP message optimization: Cisco Unified IP Phone 7935/7936. This feature may require a phone reset after update.</p>
Require off-premise location	Check this check box to allow CTI device be available at an off-premise locations.
Call Routing Information	
Inbound/Outbound Calls Information	
Calling Party Transformation CSS	This setting allows you to localize the calling party number on the device. Make sure that the Calling Party Transformation CSS that you choose contains the calling party transformation pattern that you want to assign to this device.
Use Device Pool Calling Party Transformation CSS	To use the Calling Party Transformation CSS that is configured in the device pool that is assigned to this device, check this check box. If you do not check this check box, the device uses the Calling Party Transformation CSS that you configured in the Trunk Configuration window.
Protocol Specific Information	

Field	Description
Packet Capture Mode	<p>This setting exists for troubleshooting encryption only; packet capturing may cause high CPU usage or call-processing interruptions. Choose one of the following options from the drop-down list box:</p> <ul style="list-style-type: none"> • None—This option, which serves as the default setting, indicates that no packet capturing is occurring. After you complete packet capturing, configure this setting. • Batch Processing Mode—Cisco Unified CM writes the decrypted or nonencrypted messages to a file, and the system encrypts each file. On a daily basis, the system creates a new file with a new encryption key. Cisco Unified CM, which stores the file for seven days, also stores the keys that encrypt the file in a secure location. Cisco Unified CM stores the file in the PktCap virtual directory. A single file contains the time stamp, source IP address, source IP port, destination IP address, packet protocol, message length, and the message. The TAC debugging tool uses HTTPS, administrator username and password, and the specified day to request a single encrypted file that contains the captured packets. Likewise, the tool requests the key information to decrypt the encrypted file. <p>For more information on packet capturing, see the <i>Troubleshooting Guide for Cisco Unified Communications Manager</i>.</p>
Packet Capture Duration	<p>This setting exists for troubleshooting encryption only; packet capturing may cause high CPU usage or call-processing interruptions.</p> <p>This field specifies the maximum number of minutes that is allotted for one session of packet capturing. The default setting equals 0, although the range exists from 0 to 300 minutes.</p> <p>To initiate packet capturing, enter a value other than 0 in the field. After packet capturing completes, the value, 0, displays.</p> <p>For more information on packet capturing, see the <i>Cisco Unified Communications Manager Troubleshooting Guide</i>.</p>
Presence Group	<p>Configure this field with the Presence feature.</p> <p>Note If you are not using this application user with presence, leave the default (None) setting for presence group.</p> <p>From the drop-down list box, choose a Presence group for the application user. The group selected specifies the destinations that the application user, such as IPMASysUser, can monitor.</p>

Field	Description
SIP Dial Rules	<p>If required, choose the appropriate SIP dial rule. SIP dial rules provide local dial plans for Cisco Unified IP Phones 7905, 7912, 7940, and 7960, so users do not have to press a key or wait for a timer before the call gets processed.</p> <p>Leave the SIP Dial Rules field set to <None> if you do not want dial rules to apply to the IP phone that is running SIP. This means that the user must use the Dial softkey or wait for the timer to expire before the call gets processed.</p>
MTP Preferred Originating Codec	<p>From the drop-down list box, choose the codec to use if a media termination point is required for SIP calls.</p>
Device Security Profile	<p>Choose the security profile to apply to the device.</p> <p>You must apply a security profile to all phones that are configured in Cisco Unified Communications Manager Administration. Installing Cisco Unified Communications Manager provides a set of predefined, nonsecure security profiles for auto-registration. 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.</p> <p>To identify the settings that the profile contains, choose System > Security Profile > Phone Security Profile.</p> <p>Note The CAPF settings that are configured in the profile relate to the Certificate Authority Proxy Function settings that display in the Phone Configuration window. You must configure CAPF settings for certificate operations that involve manufacturer-installed certificates (MICs) or locally significant certificates (LSC). See the Cisco Unified Communications Manager Security Guide for more information about how CAPF settings that you update in the phone configuration window affect security profile CAPF settings.</p>
Rerouting Calling Search Space	<p>From the drop-down list box, choose a calling search space to use for rerouting.</p> <p>The rerouting calling search space of the referrer gets used to find the route to the refer-to target. When the Refer fails due to the rerouting calling search space, the Refer Primitive rejects the request with the “405 Method Not Allowed” message.</p> <p>The redirection (3xx) primitive and transfer feature also uses the rerouting calling search space to find the redirect-to or transfer-to target.</p>

Field	Description
SUBSCRIBE Calling Search Space	<p>Supported with the Presence feature, the SUBSCRIBE calling search space determines how Cisco Unified Communications Manager routes presence requests that come from the end user. This setting allows you to apply a calling search space separate from the call-processing search space for presence (SUBSCRIBE) requests for the end user.</p> <p>From the drop-down list box, choose the SUBSCRIBE calling search space to use for presence requests for the end user. All calling search spaces that you configure in Cisco Unified Communications Manager Administration display in the SUBSCRIBE Calling Search Space drop-down list box.</p> <p>If you do not select a different calling search space for the end user from the drop-down list, the SUBSCRIBE calling search space defaults to None.</p> <p>To configure a SUBSCRIBE calling search space specifically for this purpose, you configure a calling search space as you do all calling search spaces.</p>
SIP Profile	<p>Choose the default SIP profile or a specific profile that was previously created. SIP profiles provide specific SIP information for the phone such as registration and keepalive timers, media ports, and do not disturb control.</p>
Digest User	<p>Choose an end user that you want to associate with the phone for this setting that is used with digest authentication (SIP security).</p> <p>Ensure that you configured digest credentials for the user that you choose, as specified in the End User Configuration window.</p> <p>For more information on digest authentication, see the <i>Cisco Unified Communications Manager Security Guide</i>.</p>
Media Termination Point Required	<p>Use this field to indicate whether a media termination point is used to implement features that H.323 does not support (such as hold and transfer).</p> <p>Check the Media Termination Point Required check box if you want to use an MTP to implement features. Uncheck the Media Termination Point Required check box if you do not want to use an MTP to implement features.</p> <p>Use this check box only for H.323 clients and those H.323 devices that do not support the H.245 empty capabilities set or if you want media streaming to terminate through a single source.</p> <p>If you check this check box to require an MTP and this device becomes the endpoint of a video call, the call will be audio only.</p>
Unattended Port	<p>Check this check box to indicate an unattended port on this device.</p>

Field	Description
Require DTMF Reception	<p>For phones that are running SIP and SCCP, check this check box to require DTMF reception for this phone.</p> <p>Note In configuring Cisco Unified Mobility features, when using intercluster DNs as remote destinations for an IP phone via SIP trunk (either intercluster trunk [ICT] or gateway), check this check box so that DTMF digits can be received out of band, which is crucial for Enterprise Feature Access midcall features.</p>
Certification Authority Proxy Function (CAPF) Information	
Certificate Operation	<p>From the drop-down list box, choose one of the following options:</p> <ul style="list-style-type: none"> • No Pending Operation—Displays when no certificate operation is occurring (default setting). • Install/Upgrade—Installs a new or upgrades an existing locally significant certificate in the phone. • Delete—Deletes the locally significant certificate that exists in the phone. • Troubleshoot—Retrieves the locally significant certificate (LSC) or the manufacture installed certificate (MIC), so you can view the certificate credentials in the CAPF trace file. If both certificate types exist in the phone, Cisco Unified CM creates two trace files, one for each certificate type. <p>By choosing the Troubleshooting option, you can verify that an LSC or MIC exists in the phone.</p> <p>For more information on CAPF operations, see the <i>Cisco Unified Communications Manager Security Guide</i>.</p>

Field	Description
Authentication Mode	<p>This field allows you to choose the authentication method that the phone uses during the CAPF certificate operation.</p> <p>From the drop-down list box, choose one of the following options:</p> <ul style="list-style-type: none"> • By Authentication String—Installs/upgrades, deletes, or troubleshoots a locally significant certificate only when the user enters the CAPF authentication string on the phone. • By Null String— Installs/upgrades, deletes, or troubleshoots a locally significant certificate without user intervention. This option provides no security; Cisco strongly recommends that you choose this option only for closed, secure environments. • By Existing Certificate (Precedence to LSC)—Installs/upgrades, deletes, or troubleshoots a locally significant certificate if a manufacture-installed certificate (MIC) or locally significant certificate (LSC) exists in the phone. If a LSC exists in the phone, authentication occurs via the LSC, regardless whether a MIC exists in the phone. If a MIC and LSC exist in the phone, authentication occurs via the LSC. If a LSC does not exist in the phone, but a MIC does exist, authentication occurs via the MIC. Before you choose this option, verify that a certificate exists in the phone. If you choose this option and no certificate exists in the phone, the operation fails. At any time, the phone uses only one certificate to authenticate to CAPF even though a MIC and LSC can exist in the phone at the same time. If the primary certificate, which takes precedence, becomes compromised for any reason, or, if you want to authenticate via the other certificate, you must update the authentication mode. • By Existing Certificate (Precedence to MIC)—Installs, upgrades, deletes, or troubleshoots a locally significant certificate if a LSC or MIC exists in the phone. If a MIC exists in the phone, authentication occurs via the MIC, regardless whether a LSC exists in the phone. If a LSC exists in the phone, but a MIC does not exist, authentication occurs via the LSC. Before you choose this option, verify that a certificate exists in the phone. If you choose this option and no certificate exists in the phone, the operation fails. <p>Note The CAPF settings that are configured in the Phone Security Profile window interact with the CAPF parameters that are configured in the Phone Configuration window.</p>

Field	Description
Authentication String	<p>If you chose the By Authentication String option in the Authentication Mode drop-down list box, this field applies. Manually enter a string or generate a string by clicking the Generate String button. Ensure that the string contains 4 to 10 digits.</p> <p>To install, upgrade, delete, or troubleshoot a locally significant certificate, the phone user or administrator must enter the authentication string on the phone.</p>
Key Size (Bits)	<p>For this setting that is used for CAPF, choose the key size for the certificate from the drop-down list box. The default setting equals 1024. Other options include 512 and 2048.</p> <p>If you choose a higher key size than the default setting, the phones take longer to generate the entropy that is required to generate the keys. Key generation, which is set at low priority, allows the phone to function while the action occurs. Depending on the phone model, you may notice that key generation takes up to 30 or more minutes to complete.</p> <p>Note The CAPF settings that are configured in the Phone Security Profile window interact with the CAPF parameters that are configured in the Phone Configuration window.</p>
Operation Completes By	<p>This field, which supports the Install/Upgrade, Delete, and Troubleshoot Certificate Operation options, specifies the date and time in which you must complete the operation.</p> <p>The values that display apply for the Unified Communications Manager publisher node.</p>
Certificate Operation Status	<p>This field displays the progress of the certificate operation; for example, <operation type> pending, failed, or successful, where operating type equals the Install/Upgrade, Delete, or Troubleshoot Certificate Operation options. You cannot change the information that displays in this field.</p>
Enable Extension Mobility	
Enable Extension Mobility	<p>Check this check box if this phone supports extension mobility.</p>
Log Out Profile	<p>This drop-down list box specifies the device profile that the device uses when no one is logged in to the device by using Cisco Extension Mobility. You can choose either Use Current Device Settings or one of the specific configured profiles that are listed.</p> <p>If you select a specific configured profile, the system retains a mapping between the device and the login profile after the user logs out. If you select Use Current Device Settings, no mapping gets retained.</p>

Field	Description
Log in Time	This field remains blank until a user logs in. When a user logs in to the device by using Cisco Extension Mobility, the time at which the user logged in displays in this field.
Log out Time	This field remains blank until a user logs in. When a user logs in to the device by using Cisco Extension Mobility, the time at which the system will log out the user displays in this field.
MLPP Information	
MLPP Domain	Choose an MLPP domain from the drop-down list box for the MLPP domain that is associated with this device. If you leave the None value, this device inherits its MLPP domain from the value that was set for the device pool of the device. If the device pool does not have an MLPP domain setting, this device inherits its MLPP domain from the value that was set for the MLPP Domain Identifier enterprise parameter.
Do Not Disturb	
Do Not Disturb	Check this check box to enable Do Not Disturb on the remote device.
DND Option	When you enable DND on the phone, Ringer Off parameter turns off the ringer, but incoming call information gets presented to the device, so the user can accept the call.
Product Specific Configuration Layout Information	
Video Capabilities	When enabled, indicates that the device will participate in video calls. Default: Enabled

You can view the directory numbers that are assigned to the phone from the Association Information area of the Phone Configuration window. After you add a phone, the Association Information area displays on the left side of the Phone Configuration window.

Table 4: Association Information Settings

Field	Description
Modify Button Items	<p>After you add a phone, the Association Information area displays on the left side of the Phone Configuration window.</p> <p>Click this button to manage button associations for this phone. A dialog box warns that any unsaved changes to the phone may be lost. If you have saved any changes that you made to the phone, click OK to continue. The Reorder Phone Button Configuration window displays for this phone.</p> <p>See the Modifying Phone Button Template Button Items topic for a detailed procedure.</p>
Line [1] - Add a new DN Line [2] - Add a new DN	<p>After you add a phone, the Association Information area displays on the left side of the Phone Configuration window.</p> <p>Click these links to add a directory number(s) that associates with this phone. When you click one of the links, the Directory Number Configuration window displays.</p> <p>See the Directory Number Configuration Settings section for details.</p>

Cisco IP Communicator

Cisco IP Communicator, a software-based application, allows users to place and receive phone calls by using their personal computers. Cisco IP Communicator depends upon the Cisco Unified Communications Manager call-processing system to provide telephony features and voice-over-IP capabilities.

This interaction with Cisco Unified Communications Manager means that Cisco IP Communicator provides the same functionality as a full-featured Cisco Unified IP Phone, while providing the portability of a desktop application. Additionally, it means that you administer Cisco IP Communicator as a phone device by using the Cisco Unified Communications Manager Administration Phone Configuration window.

Cisco Unified Personal Communicator

Cisco Unified Personal Communicator, a desktop software application, provides access to voice, video, document-sharing, and presence applications - all from a single, rich media interface. Cisco Unified Personal Communicator relies on the Cisco Unified Communications Manager call-processing system to provide telephony features and voice-over-IP capabilities.

This interaction with Cisco Unified Communications Manager enables Cisco Unified Personal Communicator to offer integrated softphone capabilities and control of the physical IP phone of the user. Additionally, it

means you administer Cisco Unified Personal Communicator as a phone device by using the Cisco Unified Communications Manager Administration Phone Configuration window.

Cisco TelePresence

The Cisco TelePresence Meeting Solution, a visual meeting room solution that comprises endpoints, IP telephony infrastructure technology, and user software applications, enables life-size, “you are there” video teleconferencing. The Cisco TelePresence IP Phone represents an integral part of the solution that provides the user interface for making connections to other Cisco TelePresence meeting rooms and for driving the codec, the device that manages the plasma display screens, microphones, speakers, and cameras that create the virtual meeting experience. The Cisco TelePresence IP Phone offers both standard Cisco Unified IP Phone 7975 and Cisco TelePresence meeting connection functionality. As an example, the Cisco TelePresence IP Phone user interface displays a schedule of the meetings for the day and provides softkeys that are designed to enable and enhance the teleconference connections but then can be used during the video teleconference to add audio meeting participants or to make voice calls.

For more information about Cisco TelePresence, see the following system and configuration documentation:

- Cisco TelePresence System Administrators Guide
- Cisco TelePresence Meeting User’s Guide
- Cisco Unified Communications Manager and Cisco TelePresence Configuration

Cisco Unified Mobile Communicator

Cisco Unified Mobile Communicator specifies a software application for mobile handsets that extends enterprise communications applications and services to mobile phones and smartphones. Cisco Unified Mobile Communicator streamlines the communication experience, enabling real-time collaboration across the enterprise.

To configure a Cisco Unified Mobile Communicator, choose the **Device > Phone** menu option in Cisco Unified Communications Manager Administration.

Codec Usage

Cisco Unified Communications Manager supports the Advertise G.722 Codec enterprise parameter, which determines whether Cisco Unified IP Phones advertise the G.722 codec to Cisco Unified Communications Manager. Codec negotiation involves two steps. First, the phone must advertise the supported codec(s) to Cisco Unified Communications Manager (not all phones support the same set of codecs). Second, when Cisco Unified Communications Manager gets the list of supported codecs from all phones that are involved in the call attempt, it chooses a commonly supported codec based on various factors, including the region pair setting. Valid values specify True (the specified Cisco Unified IP Phones advertise G.722 to Cisco Unified Communications Manager) or False (the specified Cisco Unified IP Phones do not advertise G.722 to Cisco Unified Communications Manager).

**Note**

The default for the Advertise G.722 Codec enterprise parameter enables G.722 on all phones in the cluster. The default value of the phone configuration Advertise G.722 Codec Product-Specific parameter uses the value that the enterprise parameter setting specifies.

The Product Specific Configuration Layout area in the Phone Configuration window supports the parameter, Advertise G.722 Codec. Use this parameter to override the enterprise parameter on an individual phone basis.

The following table indicates how the phone responds to the configuration options.

Table 5: How Phone Responds to Configuration Settings

Enterprise Parameter Setting	Phone (Product-Specific) Parameter Setting	Phone Advertises G.722
Advertise G.722 Codec Enabled	Use System Default	Yes
Advertise G.722 Codec Enabled	Enabled	Yes
Advertise G.722 Codec Enabled	Disabled	No
Advertise G.722 Codec Disabled	Use System Default	No
Advertise G.722 Codec Disabled	Enabled	Yes
Advertise G.722 Codec Disabled	Disabled	No

Cisco Unified Communications Manager supports G.722, which is a wideband codec, as well as a propriety codec simply named Wideband. Both represent wideband codecs. Wideband codecs such as G.722 provide a superior voice experience because wideband frequency response is 200 Hz to 7 kHz compared to narrowband frequency response of 300 Hz to 3.4 kHz. At 64 kb/s, the G.722 codec offers conferencing performance and good music quality.

When users use a headset that supports wideband, they experience improved audio sensitivity when the wideband setting on their phones is enabled (it is disabled by default). To access the wideband headset setting on the phone, users choose the **Settings** icon **User Preferences > Audio Preferences > Wideband Headset**. Users should check with their system administrator to be sure their phone system is configured to use G.722 or wideband. If the system is not configured for a wideband codec, they may not detect any additional audio sensitivity, even when they are using a wideband headset.

The following Cisco Unified IP Phones (both SCCP and SIP) support the wideband codec G.722 for use with a wideband headset:

- Cisco Unified IP Phone 7906G
- Cisco Unified IP Phone 7911G
- Cisco Unified IP Phone 7931G
- Cisco Unified IP Phone 7942G
- Cisco Unified IP Phone 7945G

- Cisco Unified IP Phone 7962G
- Cisco Unified IP Phone 7965G
- Cisco Unified IP Phone 7975G
- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

When you choose a G.711 or G.722 codec in Region Configuration, you are choosing the bandwidth utilization. Choosing either codec produces the same affect. When you choose either G.711 or G.722, these codecs disallow selecting codecs that have a payload greater than 64 kb/s, such as the G.722 wideband codec and Advanced Audio Codec (AAC) (when AAC uses more than one channel).

If you choose a region that is lower than G.711 or G.722, the Advertise G.722 Codec enterprise parameter gets ignored because the system does not allow G.722, G.711, AAC, and wideband.

**Tip**

Enabling the Advertise G.722 Codec parameter causes interoperability problems with call park and ad hoc conferences. When you use the enterprise parameter with features such as ad hoc conferencing and call park, change the setting to Disabled and update the device pools for the phones.

When enabled, the service parameter allows Cisco Unified IP Phones (such as 7971, 7970, 7941, 7961) to negotiate and use the G.722 codec when calls are within the same region.

If individual phone control and use of a specific codec type is required (for example, G.711), check the configuration of each phone (by using Phone Configuration) for the parameter Advertise G.722 Codec, and change the setting to Disabled. Save and reset the device.

**Note**

If the Advertise G.722 Codec enterprise parameter is set to Enabled, the administrator can override this by using the G.722 Codec Enabled service parameter. This service parameter determines whether Cisco Unified Communications Manager supports G.722 negotiation for none, some, or all devices. Valid values specify Enabled for All Devices (support G.722 for all devices), Enabled for All Devices Except Recording-Enabled Devices (support G.722 for all devices except those that have call recording enabled), or Disabled (do not support G.722 codec).

Phone Button Templates

Cisco Unified Communications Manager includes several default phone button templates. When adding phones, you can assign one of these templates to the phones or create a new template.

Creating and using templates provide a fast way to assign a common button configuration to a large number of phones. For example, if users in your company do not use the conference feature, you can create a template that reassigns this button to a different feature, such as speed dial.

To create a template, you must make a copy of an existing template and assign the template a unique name. You can make changes to the custom templates that you created, and you can change the labels of the default phone button templates. You cannot change the function of the buttons in the default templates. You can rename existing templates and modify them to create new ones, update custom templates to add or remove

features, lines, or speed dials, and delete custom templates that are no longer being used. When you update a template, the change affects all phones that use the template.

Renaming a template does not affect the phones that use that template. All Cisco Unified IP Phones that use this template continue to use this template after it is renamed.

Make sure that all phones have at least one line that is assigned to each phone. Normally, this assignment specifies button 1. Phones can have additional lines that are assigned, depending on the Cisco Unified IP Phone model. Phones also generally have several features, such as speed dial, that are assigned to the remaining buttons.

You can delete phone templates that are not currently assigned to any phone in your system if they are not the only template for a given phone model. You cannot delete a template that is assigned to one or more devices or the default template for a model (specified in the Device Defaults Configuration window). You must reassign all Cisco Unified IP Phones that are using the template that you want to delete to a different phone button template before you can delete the template.


Note

Use a copy of the standard phone button template for button assignment. The standard phone button template for any phone that supports expansion module include buttons for both the phone and the expansion module. For example, the Cisco Unified IP Phone 7965, which supports the Cisco Unified IP Phone Expansion Module 7915, includes buttons for both devices (up to 48 buttons).

Choose Dependency Records from the Related Links drop-down list box on the Phone Button Template Configuration window to view the devices that are using a particular template.

Cisco Unified Communications Manager does not directly control all features on Cisco Unified IP Phones through phone button templates. See the Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager and other phone documentation for detailed information about individual Cisco Unified IP Phone family models.

Default Phone Button Templates

Although all Cisco Unified IP Phones support similar features, you implement these features differently on various models. For example, some models configure features such as Hold or Transfer by using phone button templates; other models have fixed buttons or onscreen program keys for these features that are not configurable. Also, the maximum number of lines or speed dials that are supported differs for some phone models. These differences require different phone button templates for specific models.

Each Cisco Unified IP Phone comes with a default phone button template. You can use the default templates as is to quickly configure phones. You can also copy and modify the templates to create custom templates.

Custom templates enable you to make features available on some or all phones, restrict the use of certain features to certain phones, configure a different number of lines or speed dials for some or all phones, and so on, depending on how the phone will be used. For example, you may want to create a custom template that can be applied to phones that will be used in conference rooms.

The following table provides descriptions of the standard phone button templates.

Table 6: Default Phone Button Templates Listed by Model

Phone Button Template Name	Template Description

Phone Button Template Name	Template Description
Standard 7985	The Standard 7985 template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys on the Cisco IP Video Phone 7985.
Standard 7971 SCCP	The Standard 7971 SCCP template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7971.
Standard 7971 SIP	The Standard 7971 SIP template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7971.
Standard 7970 SCCP	The Standard 7970 SCCP template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7970.
Standard 7970 SIP	The Standard 7970 SIP template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7970.
Standard 7961 SCCP and Standard 7961G-GE SCCP	The Standard 7961 SCCP template uses buttons 1 and 2 for lines and assigns buttons 3 through 6 as speed dials or lines or for the features privacy and service URL. Access other phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7961.
Standard 7961 SIP	The Standard 7961 SIP template uses buttons 1 and 2 for lines and assigns buttons 3 through 6 as speed dials or lines or for the features privacy and service URL. Access other phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7961.

Phone Button Template Name	Template Description
Standard 7960 SCCP and Standard 7960 SIP	The Standard 7960 SCCP and SIP templates use buttons 1 and 2 for lines and assigns buttons 3 through 6 as speed dials or lines or for the features privacy and service URL. Access other phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7960.
Standard 7960 SIP	The Standard 7960 SIP template uses buttons 1 and 2 for lines and assigns buttons 3 through 6 as speed dials or lines or for the features privacy and service URL. Access other phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7960.
Standard 7941 SCCP and Standard 7941G-GE SCCP	The Standard 7941 SCCP template comes with a preconfigured one-line phone button template (button 1 for line 1 and button 2 for speed dial). Access phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7941.
Standard 7941 SIP	The Standard 7940 SIP template comes with a preconfigured one-line phone button template (button 1 for line 1 and button 2 for speed dial). Access phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7941.
Standard 7940 SCCP and Standard 7940 SIP	The Standard 7940 SCCP templates comes with a preconfigured one-line phone button template (button 1 for line 1 and button 2 for speed dial). Access phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7940.
Standard 7940 SIP	The Standard 7940 SIP template comes with a preconfigured one-line phone button template (button 1 for line 1 and button 2 for speed dial). Access phone features, such as abbreviated dial, call park, call forward, redial, hold, resume, call back, conferencing, and so on, by using softkeys on the Cisco Unified IP Phone 7940.
Standard 7931 SCCP and Standard 7931 SIP	The Standard 7931 SCCP and SIP templates use button 1 for line 1.
Standard 7920	The Standard 7920 template uses buttons 1 and 2 for lines and assigns buttons 3 through 6 for speed dials.

Phone Button Template Name	Template Description
Standard 7912 SCCP	The Standard 7912 SCCP template uses button 1 for line 1, buttons 2 through 5 for speed dial, button 6 for Hold, and button 7 for Settings.
Standard 7912 SIP	The Standard 7912 SIP template uses button 1 for line 1, buttons 2 through 5 for speed dial, button 6 for Hold, and button 7 for Settings.
Standard 7911 SCCP and Standard 7911 SIP	The Standard 7911 SCCP and SIP templates use button 1 for line 1, makes button 2 configurable as the Privacy softkey (default specifies None), and assigns buttons 3 through 6 as speed dials. The user accesses speed dials from the Directories menu or the Navigation button on the phone.
Standard 7911 SIP	The Standard 7911 SIP template uses button 1 for line 1, makes button 2 configurable as the Privacy softkey (default specifies None), and assigns buttons 3 through 6 as speed dials. The user accesses speed dials from the Directories menu or the Navigation button on the phone.
Standard 7910	The Standard 7910 template uses button 1 for message waiting, button 2 for conference, button 3 for forwarding, buttons 4 and 5 for speed dial, and button 6 for redial. The Cisco Unified IP Phone 7910 includes fixed buttons for Line, Hold, Transfer, and Settings.
Standard 7906 SCCP and Standard 7906 SIP	The Standard 7906 SCCP and SIP templates use button 1 for line 1, makes button 2 configurable as the Privacy softkey (default specifies None), and assigns buttons 3 through 6 as speed dials. The user accesses speed dials from the Directories menu or the Navigation button on the phone.
Standard 7906 SIP	The Standard 7906 SIP template uses button 1 for line 1, makes button 2 configurable as the Privacy softkey (default specifies None), and assigns buttons 3 through 6 as speed dials. The user accesses speed dials from the Directories menu or the Navigation button on the phone.
Standard 7905 SCCP	The Standard 7905 SCCP template uses button 1 for line 1, buttons 2 through 5 for speed dial, button 6 for Hold, and button 7 for Settings.
Standard 7905 SIP	The Standard 7905 SIP template uses button 1 for line 1, buttons 2 through 5 for speed dial, button 6 for Hold, and button 7 for Settings.
Standard 7902	The Standard 7902 template uses button 1 for line 1, buttons 2 through 5 for speed dial, button 6 for Hold, and button 7 for Settings.
Standard 7936	The Standard 7936 template, which is not configurable for the Cisco Unified IP Conference Station 7936, uses button 1 for line 1.

Phone Button Template Name	Template Description
Standard 7935	The Standard 7935 template, which is not configurable for the Cisco IP Conference Station 7935, uses button 1 for line 1.
Standard 30 SP+	<p>The Standard 30 SP+ template uses buttons 1 through 4 for lines, button 5 for call park, buttons 6 through 8 and 17 through 21 remain undefined, and buttons 9 through 13 and 22 through 25 apply for speed dial; button 14 applies for message-waiting indicator, button 15 for forward, and button 16 for conference.</p> <p>Note For only the Cisco IP Phone 30 SP+, assign button 26 for automatic echo cancellation (AEC).</p>
Standard 30 VIP	The Standard 30 VIP template uses buttons 1 through 4 for lines, button 5 for call park, buttons 6 through 13 and 22 through 26 for speed dial, button 14 for message-waiting indicator, button 15 for call forward, and button 16 for conference.
Standard 12 Series, including the 12 S, 12 SP, and 12 SP+	The Standard 12 S, Standard 12 SP, and Standard 12 SP + templates use buttons 1 and 2 for lines, button 3 for redial, buttons 4 through 6 for speed dial, button 7 for hold, button 8 for transfer, button 9 for forwarding, button 10 for call park, button 11 for message waiting, and button 12 for conference.
Default VGC Virtual Phone	The Default VGC Virtual Phone template for the Cisco VGC Virtual Phone uses button 1 for line 1.
Standard Analog	The Standard Analog template for analog phones uses button 1 for line 1.
Standard ATA 186	The Standard ATA 186 template for the Cisco ATA 186 Analog Telephone Adaptor uses button 1 for a line and buttons 2 through 10 for speed dials.
ISDN BRI Phone	The ISDN BRI Phone template uses button 1 for line 1.
Standard CIPC SCCP	The Standard CIPC (Cisco IP Communicator) SCCP template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys (by configuring the softkey template to the phone).

Phone Button Template Name	Template Description
Standard CIPC SIP	The Standard CIPC SIP template uses buttons 1 and 2 for lines and assigns buttons 3 through 8 as speed dials. Access other phone features, such as call park, call forward, redial, hold, resume, voice-messaging system, conferencing, and so on, by using softkeys (by configuring the softkey template to the phone).
Standard IP-STE	The Standard IP-STE template uses buttons 1 and 2 for lines.
Standard Unified Communicator SIP	The Standard Unified Communicator SIP template uses button 1 for line 1.
Standard VGC Phone	The Standard VGC Phone template for the Cisco VG248 Gateway uses button 1 for a line and buttons 2 through 10 for speed dials.
Standard Cisco TelePresence	The Standard Cisco TelePresence template, required by Cisco TelePresence, uses buttons 1 and 2 for lines and buttons 3 through 42 for speed dials.
Third-Party SIP Device (Advanced)	The Generic SIP Phone - 2 Lines template, which is used for third-party phones that run SIP, uses buttons 1 and 2 for lines.
Third-Party SIP Device (Basic)	The Generic SIP Phone - 2 Lines template, which is used for third-party phones that run SIP, uses buttons 1 and 2 for lines.
Third-Party AS-SIP Device	The Generic SIP Phone - 2 Lines template, which is used for third-party phones that run SIP, uses buttons 1 and 2 for lines.

Guidelines For Customizing Phone Button Templates

Use the following guidelines when you are creating custom phone button templates:

- Make sure that phone users receive a quick reference card or getting started guide that describes the most basic features of the custom template. If you create a custom template for employees in your

company to use, make sure that it includes the following features and that you describe them on the quick reference card that you create for your users:

- Cisco Unified IP Phone 7975, 7965, 7962, 7960, 7945, 7942, 7940, 7911, 7906-Line (one or more)
- Cisco VGC Virtual Phone and Cisco ATA 186-Line and speed dials
-
-
- Consider the nature of each feature to determine how to configure your phone button template. You may want to assign multiple buttons to speed dial and line; however, you usually require only one of the other phone button features that are described in Table 36-6.

Table 7: Phone Button Feature Description

Feature	Description
AEC	If you are configuring a template for the Cisco IP Phone 30 VIP, you must include one occurrence of this feature and assign it to button 26. Auto echo cancellation (AEC) reduces the amount of feedback that the called party receives when the calling party is using a speakerphone. Users should press the AEC button on a Cisco IP Phone 30 SP+ when they are using speakerphone. Users do not need to press this button when speakerphone is not in use. This feature requires no configuration for it to work.
Answer/release	In conjunction with a headset apparatus, the user can press a button on the headset apparatus to answer and release (disconnect) calls.
Auto answer	If this feature is programmed on the template, pressing this button causes the speakerphone to go off hook automatically when an incoming call is received. Note You configure this feature for some phones models by using the Phone Button Template window, and you configure this feature for some phone models by using the Phone Configuration window.
Call park	In conjunction with a call park number or range, when the user presses this button, call park places the call at a directory number for later retrieval. You must have a call park number or range that is configured in the system for this button to work, and you should provide that number or range to your users, so they can dial in to the number(s) to retrieve calls.

Feature	Description
Call Park BLF	<p>Users can monitor the busy/idle status of directed call park numbers using the Call Park Busy Lamp Field (BLF) buttons. Users can also speed dial those numbers by pressing the BLF buttons. One directed call park number gets configured for each Call Park BLF button. To successfully park or retrieve a call by using a Call Park BLF button, you must ensure that the partition and the calling search space of the device are configured correctly.</p> <p>Note Use this button with the directed call park feature (a transfer function), not with the standard call park feature (a hold function).</p>
Conference	<p>Users can initiate an ad hoc conference and add participants by pressing the Conference button. (Users can also use the Join softkey to initiate an ad hoc conference.)</p> <p>Only the person who initiates an ad hoc conference needs a conference button. You must make sure that an ad hoc conference bridge device is configured in Cisco Unified Communications Manager Administration for this button to work. See the Conference Bridges chapter for more information.</p>
Forward all	<p>Users press this button to forward all calls to the designated directory number. Users can designate forward all in the Cisco Unified IP Phone Configuration windows, or you can designate a forward all number for each user in Cisco Unified Communications Manager Administration.</p>
Hold	<p>Users press this button to place an active call on hold. To retrieve a call on hold, users press the flashing line button or lift the handset and press the flashing line button for the call on hold. The caller on hold receives a tone every 10 seconds to indicate the hold status or music (if the Music On Hold feature is configured). The hold tone feature requires no configuration to work.</p>
Line	<p>Users press this button to dial a number or to answer an incoming call. For this button to work, you must have added directory numbers on the user phone.</p>

Feature	Description
Meet-Me conference	When users press this button, they initiate a meet-me conference, and they expect other invited users to dial in to the conference. Only the person who initiates a meet-me conference needs a meet-me button. You must make sure that a meet-me conference device is configured in Cisco Unified Communications Manager Administration for this button to work.
Message waiting	Users press this button to connect to the voice-messaging system.
None	Use None to leave a button unassigned.
Privacy	Users press this button to activate/deactivate privacy.
Redial	Users press this button to redial the last number that was dialed on the Cisco Unified IP Phone. This feature requires no configuration to work.
Service URL	Users press this button to access a Cisco Unified IP Phone Service such as personal fast dials, stock quotes, or weather.
Speed Dial	Users press this button to speed dial a specified number. System administrators can designate speed-dial numbers in Cisco Unified Communications Manager Administration. Users can designate speed-dial numbers in the Cisco Unified Communications Self Care Portal.
Speed Dial/BLF	Users monitor this button for the real-time status of the associated directory number or SIP URI on those devices that support the presence feature. Users press this button to dial the destination.
Transfer	Users press this button to transfer an active call to another directory number. This feature requires no configuration to work.

Programmable Line Keys

Cisco Unified IP Phones support line buttons (the buttons next to the phone screen), which are used to initiate, answer, or switch to a call on a particular line. A limited number of features, such as speed dial, extension mobility, privacy, BLF speed dial, DND, and Service URLs, get assigned to these buttons.

The Programmable Line Key (PLK) feature expands the list of features that can be assigned to the line buttons to include features that softkeys normally control; for example, New Call, Call Back, End Call, and Forward All. When you configure these features on the line buttons, they always remain visible, so you can have a “hard” New Call key.

Programmable line keys support up to 27 features on line buttons (see Table 36-6). Use the Phone Button Template Configuration window to assign programmable line keys. It provides the appropriate configurable feature for the phone model. After configuring the phone button template, you must assign the phone button template to the phone by using Phone Configuration (reset is required).

Table 8: Programmable Line Keys for Cisco Unified IP Phones

Feature	Phone Model 7971, 7970, 7961, 7941, 7914, 7915, 7916	Phone Model 7931 (SCCP only)
Redial	Yes	No, uses existing line button
Hold	Yes	No, uses existing line button
Transfer	Yes	No, uses existing line button
Privacy	Yes	Yes
Forward All	Yes	Yes
Meet Me	Yes	Yes
Conference	Yes	Yes
Park	Yes	Yes
Pickup	Yes	Yes
Group Call Pickup	Yes	Yes
Malicious Caller ID (MCID)	Yes	Yes
Conf List	Yes	Yes
Remove Last Participant	Yes	Yes
QRT	Yes	Yes
Call Back	Yes	Yes
Other Call Pickup	Yes	Yes
Video Mode	Yes	Yes
New Call	Yes	Yes
End Call	Yes	Yes

Feature	Phone Model 7971, 7970, 7961, 7941, 7914, 7915, 7916	Phone Model 7931 (SCCP only)
HLog (Hunt Group)	Yes	Yes
Mobility	Yes	Yes
Settings	No, uses existing button	Yes
Information	No, uses existing button	No
Services	No, uses existing button	Yes
Messages	No, uses existing button	Yes
Directories	No, uses existing button	Yes
AppMenu	No, uses existing button	Yes
Headset	No, uses existing button	Yes

The programmable line feature does not affect the existing softkey functionality. Softkeys still display as required and will continue to be specific to the state of the phone (for example, making a call, being in a call, navigating the Services menu).

If a feature is already assigned to a programmable line key, it can also appear as a softkey (and vice versa).

If a phone has a hard button for a feature, it cannot also have that feature as a programmable line key; for example, transfer cannot be a programmable line key on a Cisco Unified IP Phone 7931 because it already has a dedicated hard transfer button.

Softkey Templates

Use softkey templates to manage softkeys that are associated with applications such as Cisco Unified Communications Manager Assistant or call-processing features such as Call Back on Cisco Unified IP Phones. You access the Softkey Template Configuration windows in Cisco Unified Communications Manager Administration to create and update softkey templates. (**Device > Device Settings > Softkey Templates**)

Cisco Unified Communications Manager supports two types of softkey templates: standard and nonstandard. Standard softkey templates in the Cisco Unified Communications Manager database contain the recommended selection and positioning of the softkeys for an application. Cisco Unified Communications Manager provides the following standard softkey templates:

- Standard User
- Standard Chaperone Phone
- Standard Feature
- Standard Assistant
- Standard Protected Phone

- Standard Shared Mode Manager
- Standard Manager

**Note**

For most Cisco Unified IP Phone models, such as the Cisco Unified IP Phone 7945, 7965, 7975, and so on, you must assign standard or nonstandard softkey templates to the Cisco Unified IP Phone by assigning the templates individually to each phone or by assigning the common device configuration to each phone. Some Cisco Unified IP Phone models, such as the Cisco Unified IP Phone 8961, 9971, and 9951, do not use softkey templates. To determine whether your phone uses softkey templates and to determine which softkeys are supported on your phone, see the Cisco Unified IP Phone Phone Guide for your phone model.

You create a nonstandard softkey template by using the Softkey Template Configuration windows in Cisco Unified Communications Manager Administration. To create a nonstandard softkey template, the administrator copies a standard softkey template and makes changes. The administrator can add and remove applications that are associated with any nonstandard softkey template. Additionally, the administrator can configure softkey sets for each call state for a nonstandard softkey template.

The Softkey Template Configuration window lists the standard and nonstandard softkey templates and uses different icons to differentiate between standard and nonstandard templates.

The administrator assigns softkey templates in the following Cisco Unified Communications Manager Administration configuration windows:

- Common Device Configuration
- Phone Configuration (SIP and SCCP)
- UDP Template Configuration
- Default Device Profile Configuration

Add Application

You can add a standard softkey template that is associated with a Cisco application to a nonstandard softkey template. When the administrator clicks the **Add Application** button from the Softkey Template Configuration window, a separate window displays and allows you to choose the standard softkey template that is to be added to the end of the nonstandard softkey template. Duplicate softkeys get deleted from the end of the set that is moving to the front of the set.

**Tip**

To refresh the softkeys for an application in the nonstandard softkey template, choose the standard softkey template that is already associated with the nonstandard softkey template. For example, if the administrator originally copied the Standard User template and deleted some buttons, choose the Standard User softkey template by clicking on the **Add Application** button. This adds the buttons that are included in the chosen softkey template.

The number of softkeys in any given call state cannot exceed 16. A message displays, and the add application procedure stops when the maximum number of softkeys is reached. The administrator must manually remove some softkeys from the call state before trying to add another application to the template.

The **Remove Application** button allows you to delete application softkey templates that are associated with a nonstandard softkey template. Only the softkeys that are associated with the application get deleted. When

softkeys are commonly shared between applications, they remain in the softkey template until the last application that shares the softkeys is removed from the softkey template.

Configure Softkey Layout

The administrator can configure softkey sets for each call state for a nonstandard softkey template. When the administrator chooses Configure Softkey Layout from the Related Links drop-down list box on the Softkey Template Configuration window and clicks Go, the Softkey Layout Configuration window displays.

The Softkey Layout Configuration window allows you to specify the softkeys and their relative order for any phone models that support downloadable softkey templates. This window lists all softkeys, even though some phone models do not support all softkeys. To determine whether your phone model supports a softkey, see the Cisco Unified IP Phone Phone Guide for your phone model. If you choose a softkey that is not supported by the phone, the softkey does not display on the phone, even if you add it to the Selected Softkeys pane.



Note

Cisco recommends that a softkey remain in the same position for each call state. This provides the user with consistency and ease of use; for example, the More softkey always appears in the fourth softkey position from the left for each call state.

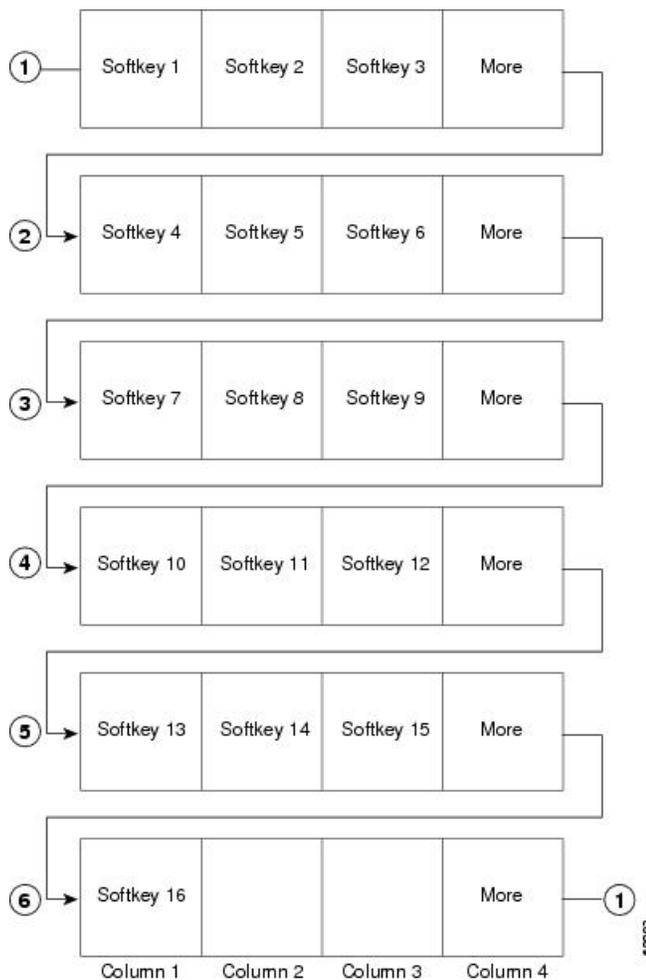
The Softkey Layout Configuration pane contains the following fields:

- Select a call state to configure—This drop-down list box displays the different call states of a Cisco Unified IP Phone. You cannot add, update, or delete call states. The call state that gets chosen from the drop-down list box indicates the softkeys that are available for that call state. Table 36-8 lists the call states.

Call State	Description
Connected	Displays when call is connected
Connected Conference	Consultation call for conference in connected call state
Connected Transfer	Consultation call for transfer in connected call state
Digits After First	Off-hook call state after user enters the first digit
Off Hook	Dial tone presented to phone
Off Hook With Feature	Off-hook call state for transfer or conference consultation call
On Hold	Call on hold
On Hook	No call exists for that phone.
Remote In Use	Another device that shares the same line uses call.
Ring In	Call received and ringing
Ring Out	Call initiated and the destination ringing

- Unselected Softkeys-Lists softkeys that are associated with a call state. This field lists the unselected, optional softkeys of the call state that displays in the Select a Call State to Configure drop-down list box. The softkeys that are listed in this field get added to the Selected Softkeys field by using the right arrows. You can add the Undefined softkey more than once to the Selected Softkey list. Choosing Undefined results in a blank softkey on the Cisco Unified IP Phone.
- Selected Softkeys-Lists softkeys that are associated with the chosen call state. This field lists the chosen softkeys of the call state that displays in the Select a Call State to Configure drop-down list box. The maximum number of softkeys in this field cannot exceed 16. See the figure which follows for a sample softkey layout.

Figure 1: Sample Softkey Layout



Softkey Template Operation

For applications such as Cisco Unified Communications Manager Assistant to support softkeys, ensure softkeys and softkey sets are configured in the database for each device that uses softkey templates and the application.

You can mix application and call-processing softkeys in any softkey template. A static softkey template associates with a device in the database. When a device registers with Cisco Unified Communications Manager, the static softkey template gets read from the database into call processing and then gets passed to the device to be used throughout the session (until the device is no longer registered or is reset). When a device resets, it may get a different softkey template or softkey layout because of updates that the administrator makes.

Softkeys support a field called application ID. An application, such as Cisco Unified Communications Manager Assistant, activates/deactivates application softkeys by sending a request to the device through the Cisco CTIManager and call processing with a specific application ID.

When a user logs in to the Cisco IP Manager Assistant service and chooses an assistant for the service, the application sends a request to the device, through Cisco CTIManager and call processing, to activate all its softkeys with its application ID.

At any time, several softkey sets may display on a Cisco Unified IP Phone (one set of softkeys for each call).

The softkey template that is associated with a device (such as a Cisco Unified IP Phone) in the database designates the one that is used when the device registers with call processing. Perform the association of softkey templates and devices by using Softkey Template configuration in Cisco Unified Communications Manager Administration.

Common Phone Profiles

Cisco Unified Communications Manager uses common phone profiles to define phone attributes that are associated with Cisco Unified IP Phones. Having these attributes in a profile instead of adding them individually to every phone decreases the amount of time that administrators spend configuring phones and allows the administrator to change the values for a group of phones. Common phone profiles specify the following attributes:

- Profile name
- Profile description
- Local phone unlock password
- DND option
- DND incoming call alert
- Phone personalization
- End user access to phone background image setting

The common phone profile remains a required field when phones are configured; therefore, you must create the common phone profile before you create a phone. Cisco Unified Communications Manager provides a Standard Common Phone Profile that you can copy and modify to create a new common phone profile. You cannot modify nor delete the Standard Common Phone Profile.

Methods for Adding Phones

You can automatically add phones that support either SCCP or SIP to the Cisco Unified Communications Manager database by using autoregistration, manually by using the phone configuration windows, or in groups with the Bulk Administration Tool (BAT).

By enabling autoregistration before you begin installing phones, you can automatically add a Cisco Unified IP Phone to the Cisco Unified Communications Manager database when you connect the phone to your IP telephony network. During autoregistration, Cisco Unified Communications Manager assigns the next available sequential directory number to the phone. In many cases, you may not want to use autoregistration; for example, if you want to assign a specific directory number to a phone or if you plan to implement authentication or encryption.



Tip Cisco Unified Communications Manager automatically disables autoregistration if you configure the clusterwide security mode for authentication and encryption through the Cisco CTL client.

If you do not use autoregistration, you must manually add phones to the Cisco Unified Communications Manager database or use the Bulk Administration Tool (BAT). BAT enables system administrators to perform batch add, modify, and delete operations on large numbers of Cisco Unified IP Phones.



Tip After you install Cisco Unified Communications Manager, if auto-registration is not enabled and the phone has not been added to the Cisco Unified Communications Manager database, the phone does not attempt to register with Cisco Unified Communications Manager. The phone continues to display the Configuring IP message until auto-registration gets enabled or until the phone gets added to the Cisco Unified Communications Manager database. The Real-Time Monitoring Tool and Cisco Unified Reporting can display information on registered and unregistered devices.

User/Phone Add

You can use the End User, Phone, DN, and LA Configuration window to add a new phone at the same time that you add a new end user. You can associate a directory number (DN) and line appearance (LA) for the new end user by using the same window. To access the End User, Phone, DN, and LA Configuration window, choose the **User Management > User/Phone Add** menu option.



Note The End User, Phone, DN, and LA Configuration window only allows addition of a new end user and a new phone. The window does not allow entry of existing end users or existing phones.

Phone Migration

The Phone Migration window in Cisco Unified Communications Manager Administration allows you to migrate feature, user, and line configuration for a phone to a different phone; that is, you can migrate data to a different phone model or to the same phone model that runs a different protocol. For example, you can migrate data from a Cisco Unified IP Phone 7965 to a Cisco Unified IP Phone 7975; or, you can migrate data from a phone model that runs SCCP, for example, the Cisco Unified IP Phone 7965 (SCCP), and move it to the same phone model that runs SIP, for example, the Cisco Unified IP Phone 7965 (SIP).



Tip Phone migration allows you to move existing phone configuration to a new phone without the need to add a phone, lines, speed dials, and so on.

Before you can migrate phone configuration to a new phone, consider the following information:

- If the phone models do not support the same functionality, be aware that you may lose functionality on the new phone. Before you save the migration configuration in the Phone Migration window, Cisco Unified Communications Manager Administration displays a warning that you may lose feature functionality.
- Some phone models do not support phone migration; for example, CTI port, H.323 client, Cisco Unified Mobile Communicator, and Cisco IP Softphone.
- Before you can migrate the phone configuration, you must create a phone template for the phone model to which you want to migrate in BAT (**Bulk Administration > Phones > Phone Template**). For example, if you want to migrate the configuration for a Cisco Unified IP Phone 7965 to a Cisco Unified IP Phone 7975, you create the phone template for the Cisco Unified IP Phone 7975.
- The new phone uses the same existing database record as the original phone, so migrating the phone configuration to the new phone removes the configuration for the original phone from Cisco Unified Communications Manager Administration/the Cisco Unified Communications Manager database; that is, you cannot view or access the configuration for the original phone after the migration.

Migrating to a phone that uses fewer speed dials or lines does not remove the speed dials or lines for the original phone from Cisco Unified Communications Manager Administration/the Cisco Unified Communications Manager database, although some of the speed dials/lines do not display on the new phone. After you migrate the configuration, you can see all speed dials and lines for the original phone in the Phone Configuration window for the new phone.

- Before you migrate the phone configuration to a new phone, ensure that the phones are unplugged from the network. After you perform the migration tasks, you can plug the new phone into the network.
- Before you migrate the phone configuration to a new phone, ensure that you have enough device license units for the new phone.
- If you want to migrate the configuration for multiple phones, use the Bulk Administration Tool.

Phone Features

Cisco Unified Communications Manager enables you to configure the following phone features on Cisco Unified IP Phones: barge, privacy release, call back, call park, call pickup, immediate divert, join across lines, malicious call identification, quality report tool, service URL, single button barge/cbarge, and speed dial and abbreviated dial.

Agent Greeting

Agent Greeting enables Cisco Unified Communications Manager to automatically play a pre-recorded announcement following a successful media connection to the agent device. The greeting helps keep agents sounding fresh because they do not have to repeat common phrases on each call. Agent Greeting is audible for the agent and the customer.

If you want to use agent greeting, Built-in Bridge must be On.

Audible Message Waiting Indicator (AMWI)

You can configure Cisco Unified IP Phones, so if voice messages are waiting, the end users will receive a stutter dial tone when the phone goes off hook (on the line on which the voice message has been left) by setting the Audible Message Waiting Indicator Policy service parameter in Cisco Unified Communications Manager Administration.

**Note**

To ensure backward compatibility, the Cisco Unified IP Phones that are running SCCP will not issue the AMWI stutter dial-tone for phones that are using SCCP firmware versions older than 10. This remains true regardless whether the AMWI is configured on the Cisco Unified Communications Manager Administration window.

Barge and Privacy

The Barge and Privacy features work together. Both features work with phones that run SIP or SCCP by using only shared lines.

Barge adds a user to a call that is in progress. Pressing the Barge or cBarge softkey automatically adds the user (initiator) to the shared-line call (target), and the users currently on the call receive a tone.

Privacy allows a user to allow or disallow other users of shared-line devices to view the device call information or to allow another user to barge in to its active calls.

Calling Party Normalization

In line with E.164 standards, calling party normalization enhances the dialing capabilities of some phones and improves call back functionality when a call is routed to multiple geographical locations; that is, the feature ensures that the called party can return a call without having to modify the directory number in the call log directories on the phone. Additionally, calling party normalization allows you to globalize and localize phone numbers, so the appropriate calling number presentation displays on the phone.

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

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

You can configure the international escape character, +, to globalize the calling party number.

Related Topics

[Use the International Escape Character](#)

Call Forward

Call forward allows a user to configure a Cisco Unified IP Phone, so all calls that are destined for it ring another phone. Configure call forward in the Directory Number Configuration window in Cisco Unified Communications Manager Administration.

**Tip**

You can configure each call forward type for internal and external calls and can forward calls to voice-messaging system or a dialed destination number by configuring the calling search space.

The administrator configures call forward information display options to the original dialed number or the redirected dialed number, or both. The administrator enables or disables the calling line ID (CLID) and calling name ID (CNID). The display option gets configured for each line appearance.

Call Forward All, Including CFA Destination Override, CFA Loop Prevention, and CFA Loop Breakout

Call Forward All (CFA) allows a phone user to forward all calls to a directory number.

The administrator can configure CFA for internal and external calls and can forward calls to a voice-messaging system or a dialed destination number by configuring the calling search space. Cisco Unified Communications Manager includes a secondary Calling Search Space (CSS) configuration field for Call Forward All (CFA). The secondary CSS for CFA combines with the existing CSS for CFA to allow support of the alternate CSS system configuration. When CFA is activated, only the primary and secondary CSS for CFA get used to validate the CFA destination and redirect the call to the CFA destination. If these fields are empty, the null CSS gets used. Only the CSS fields that are configured in the primary CSS for CFA and secondary CSS for CFA fields get used. If CFA is activated from the phone, the CFA destination gets validated by using the CSS for CFA and the secondary CSS for CFA, and the CFA destination gets written to the database. When a CFA is activated, the CFA destination always gets validated against the CSS for CFA and the secondary CSS for CFA.

Cisco Unified Communications Manager provides a service parameter (CFA Destination Override) that allows the administrator to override Call Forward All (CFA) when the target of the CFA calls the initiator of the CFA, so the CFA target can reach the initiator for important calls. In other words, when the user to whom calls are being forwarded (the target) calls the user whose calls are being forwarded (the initiator), the phone of the initiator rings instead of the call being forwarded back to the target. The override works whether the CFA target phone number is internal or external.

When the CFA Destination Override service parameter is set to False (the default value), no override occurs. Ensure the service parameter is set to True for CFA override to work.

**Note**

CFA override only takes place if the CFA destination matches the calling party and the CFA Destination Override service parameter is set to True. If the service parameter is set to True and the calling party does not match the CFA destination, CFA override does not take place, and the CFA remains in effect.

Cisco Unified Communications Manager prevents Call Forward All activation on the phone when a Call Forward All loop is identified. For example, Cisco Unified Communications Manager identifies a call forward loop when the user presses the CFwdALL softkey on the phone with directory number 1000 and enters 1001 as the CFA destination, and 1001 has forwarded all calls to directory number 1002, which has forwarded all calls to directory number 1003, which has forwarded all calls to 1000. In this case, Cisco Unified

Communications Manager identifies that a loop occurs and prevents CFA activation on the phone with directory number 1000.



Tip

If Call Forward All activation occurs in Cisco Unified Communications Manager Administration or the Cisco Unified Communications Self Care Portal, Cisco Unified Communications Manager does not prevent the CFA loop.



Tip

If the same directory number exists in different partitions, for example, directory number 1000 exists in partitions 1 and 2, Cisco Unified Communications Manager allows the CFA activation on the phone.

The Forward Maximum Hop Count service parameter, which supports the Cisco CallManager service, specifies the maximum number of call hops that can occur for a Call Forward All chain; for example, if the value of this parameter equals 7, and a Call Forward All chain occurs consecutively from directory numbers 1000 to 1007, which equals 7 hops, Cisco Unified Communications Manager prevents a phone user with directory number 2000 from activating CFA to directory number 1000 because no more than 7 forwarding hops are supported for a single call. For more information on this service parameter, including special considerations for calls that use Q.SIG trunks, click the Forward Maximum Hop Count link in the Service Parameter Configuration window in Cisco Unified Communications Manager Administration.

Cisco Unified Communications Manager prevents Call Forward All loops if CFA is activated from the phone, if the number of hops for a Call Forward All call exceeds the value that is specified for the Forward Maximum Hop Count service parameter, and if all phones in the forwarding chain have CFA activated [not Call Forward Busy (CFB), Call Forward No Answer (CFNA), or any other call forwarding options]. For example, if the user with directory number 1000 forwards all calls to directory number 1001, which has CFB and CFNA configured to directory number 1002, which has CFA configured to directory number 1000, Cisco Unified Communications Manager allows the call to occur because directory number 1002 acts as the CFB and CFNA (not CFA) destination for directory number 1001.

Call Forward All loops do not impact call processing because Cisco Unified Communications Manager supports CFA loop breakout, which ensures that if a CFA loop is identified, the call goes through the entire forwarding chain, breaks out of the Call Forward All loop, and completes as expected, even if CFNA, CFB, or other forwarding options are configured along with CFA for one of the directory numbers in the forwarding chain. For example, the user for the phone with directory number 1000 forwards all calls to directory number 1001, which has forwarded all calls to directory number 1002, which has forwarded all calls to directory number 1000, thus creating a CFA loop. In addition, directory number 1002 has configured CFNA to directory number 1004. The user at the phone with directory number 1003 calls directory number 1000, which forwards to 1001, which forwards to 1002. Cisco Unified Communications Manager identifies a CFA loop, and the call, which breaks out of the loop, tries to connect to directory number 1002. If the No Answer Ring Duration timer expires before the user for the phone with directory number 1002 answers the call, Cisco Unified Communications Manager forwards the call to directory number 1004.

For a single call, Cisco Unified Communications Manager may identify multiple Call Forward All loops and attempts to connect the call after each loop is identified.

Call Forward Busy

The Call Forward Busy (CFB) feature forwards calls only when the line is in use and the busy trigger setting is reached.

The call forward busy trigger gets configured for each line appearance and cannot exceed the maximum number of calls that are configured for a line appearance. The call forward busy trigger determines how many active calls exist on a line before the call forward busy setting gets activated (for example, 10 calls).



Tip Keep the busy trigger slightly lower than the maximum number of calls, so users can make outgoing calls and perform transfers.



Tip If a call gets forwarded to a directory number that is busy, the call does not complete.

Call Forward No Answer

The Call Forward No Answer (CFNA) feature forwards calls when the phone is not answered after the configured no answer ring duration timer is exceeded or if the destination is unregistered.

The call forward no answer ring duration gets configured for each line appearance, and the default specifies 12 seconds. The call forward no answer ring duration determines how long a phone rings before the call forward no answer setting gets activated.

Call Forward No Coverage

The Call Forward No Coverage feature forwards calls when ringing either exhausts or times out and the associated hunt-pilot for coverage specifies Use Personal Preferences for its final forwarding.

Call Waiting

Call waiting feature lets users receive a second incoming call on the same line without disconnecting the first call. When the second call arrives, the user receives a brief call-waiting indicator tone, which is configured with the Ring Setting (Phone Active) in the Directory Number Configuration window.

Configure call waiting in the Directory Number Configuration window in Cisco Unified Communications Manager Administration by setting the busy trigger (greater than 2) and maximum number of calls.

Cancel Call Waiting

The Cancel Call Waiting feature allows the user to cancel the call waiting service when a call is active. This feature enables the user to block the operation of call waiting for one call. To invoke this feature, the user dials the cancel call waiting code, obtains recall dial tone, and places a call normally. During this call, the Call Waiting service is rendered inactive, so that anyone calling the user receives the normal busy treatment, and no call waiting tones interrupt the call.



Note This feature is available on both IP and analog phones.

The administrator can enable the Cancel Call Waiting feature through a Cancel Call Waiting softkey in Cisco Unified Communications Manager, which adds a new softkey to non-standard softkey templates. The administrator then assigns the template to supported devices.

**Note**

For more information on softkey templates, see the [Softkey Templates](#), on page 61.

Call Diagnostics and Voice-Quality Metrics

You can configure Cisco Unified IP Phones that are running SCCP and SIP to collect call diagnostics and voice-quality metrics by setting the Call Diagnostics Enabled service parameter in Cisco Unified Communications Manager Administration.

SIP fully supports Call Diagnostics and Voice Quality Metrics on Cisco Unified IP Phones. Support includes end-of-call reporting, midcall reporting (for example, call hold, media disconnect), and voice quality metrics. Cisco Unified IP Phones 7940 and 7960 that are running SIP do not report voice quality metrics or midcall reporting. To enable voice quality metrics on Cisco Unified IP Phones for SIP, check the Call Stats check box on the SIP Profile Configuration window.

Call Park

Call park allows a user to place a call on hold, so anyone who is configured to use call park on the Cisco Unified Communications Manager system can retrieve it.

For example, if a user is on an active call at extension 1000, the user can park the call to a call park extension such as 1234, and another user can dial 1234 to retrieve the call.

To use call park, you must add the call park extension (in this case, 1234) in Cisco Unified Communications Manager Administration when you are configuring phone features.

Call Pickup

Cisco Unified Communications Manager provides the following types of call pickup:

- Call pickup-Allows you to answer a ringing phone in your designated call pickup group.
- Group call pickup-Allows you to answer incoming calls in another pickup group.
- Other group pickup-Allows you to answer incoming calls in a pickup group that is associated with your own group.
- Directed call pickup-Allows you to answer incoming calls directly on a specific directory number (DN) that belongs to a pickup group that is associated with your own group.

All types of call pickup can operate automatically or manually. If the service parameter, Auto Call Pickup Enabled, is enabled, Cisco Unified Communications Manager automatically connects you to the incoming call after you press one of the following softkeys on the phone:

- Pickup-For call pickup (calls in your own pickup group)
- GPickUp-For group call pickup (calls in another pickup group) and directed call pickup (calls in a pickup group that is associated with your own pickup group)
- OPickUp-For other group pickup (calls in a pickup group that is associated with your own pickup group)

After the call pickup feature is automated, you need to use only one keystroke for a call connection except for group call pickup and directed call pickup. For group call pickup, you press the GPickUp softkey on the phone and dial the DN of the other pickup group. For directed call pickup, you press the GPickUp softkey on the phone and dial the DN of the ringing phone that you want to pick up.

**Note**

CTI applications support monitoring of the party whose call is picked up. CTI applications do not support monitoring of the pickup requester or the destination of the call that is picked up. Hence, Cisco Unified Communications Manager Assistant does not support auto call pickup (one-touch call pickup).

You configure the call pickup feature when you are configuring phone features in Cisco Unified Communications Manager.

When you are adding a line, you can indicate the call pickup group. The call pickup group indicates a number that can be dialed to answer calls to this directory number (in the specified partition).

Call Pickup Notification

This feature allows users to receive an audio and/or visual alert when a call rings on a phone in pickup groups in which they are a member. For multiple-line phones, be aware that the alert is available for pickup groups that are associated with the primary line only.

You can configure the following notification parameters in the Call Pickup Group Configuration window:

- Type of notification (audio, visual, both, or neither)
- Content of the visual notification message (called party identification, calling party identification, both, or neither)
- Number of seconds delay between the time the call comes into the original called party and the notification to the rest of the call pickup group members

In the Directory Number Configuration window, you can configure the type of audio notification that is provided when a phone is idle or in use.

Call Select

The Select softkey allows a user to select a call for feature activation or to lock the call from other devices that share the same line appearance. Pressing the Select softkey on a selected call deselects the call.

When the call gets selected by a device, it gets put in the Remote-In-Use state on all other devices that share the line appearance. No one can select a call that is in the Remote-In-Use state. In other words, selecting a call instance will lock it from other devices that share the same line appearance.

A special display symbol identifies selected calls.

Call Select supports shared lines for phones that run SIP or SCCP. Select on nonshared lines does not get supported for phones that are running SIP.

Conference Linking

Advanced ad hoc conferencing allows you to link multiple ad hoc conferences together by adding an ad hoc conference to another ad hoc conference as if it were an individual participant. Two types of conference linking exist: linear and nonlinear.

Conference List

The conference list feature provides a list of participant directory numbers that are in an ad hoc conference. The name of the participant displays if it is configured in Cisco Unified Communications Manager Administration.

Any participant can invoke the conference list feature on the phone and can view the participants. The conference controller can invoke the conference list feature and can view and remove any participant in the conference by using the Remove softkey.

**Note**

On a regular conference call, the Show Details softkey displays the participants in the conference. However, for a conference call made across the cluster, the Show Details softkey displays the message "Key is Not Active".

Connected Number Display

When a call routes through a translation or route pattern, routes to a Call Forward All or Call Forward Busy destination, or gets redirected through a call transfer or CTI application, the connected number display updates to show the modified number or redirected number.

The Connected Number Display restriction restricts the connected line ID presentation to dialed digits only for the duration of the call.

Device Mobility

Cisco Unified Communications Manager uses IP subnets and device pools that contain location information to determine a device home location. By linking IP subnets to locations, the system can determine whether a device is at its home location or a remote location and register the device accordingly.

To support device mobility, modifications to the device pool structure separate the user information from the location and mobility information. The device pool contains the information that pertains to the device itself and to device mobility. An added common profile allows you to configure all the user-related information. You must associate each device with the common profile for user based information.

Direct Transfer

Using the DirTrfr and Select softkeys, a user can transfer any two established calls to remove the calls from the IP phone. For more information about Direct Transfer, see the [Make and Receive Multiple Calls Per Directory Number](#).

Directed Call Park

Directed Call Park allows a user to transfer a parked call to an available user-selected directed call park number. Configure directed call park numbers in the new Cisco Unified Communications Manager Directed Call Park Configuration window. You can configure phones that support the directed call park Busy Lamp Field (BLF) button to monitor the busy/idle status of specific directed call park numbers. Users can also use the BLF button to speed dial a directed call park number.

A user can retrieve a parked call by dialing a configured retrieval prefix followed by the directed call park number where the call is parked.

**Note**

Cisco recommends that you treat Call Park (a hold function) and Directed Call Park (a transfer function) as mutually exclusive: enable one or the other, but not both. If you do enable both, ensure that the numbers that are assigned to each are exclusive and do not overlap.

Do Not Disturb

The Do Not Disturb (DND) feature provides the following options:

- **Call Reject**-This option specifies that no incoming call information gets presented to the user. Depending on how you configure the DND Incoming Call Alert parameter, the phone may play a beep or display a flash notification of the call.
- **Ringer Off**-This option turns off the ringer, but incoming call information gets presented to the device, so that the user can accept the call.

When DND is enabled, you can also choose to have the Cisco Unified IP Phone beep or flash to indicate an incoming call. Users can configure DND directly from their Cisco Unified IP Phone or from the Cisco Unified CM User Options window.

When DND is enabled, all new incoming calls with normal priority will honor the DND settings for the device. High-priority calls, such as calls from Cisco Emergency Responder (CER) or calls with Multi-Level Precedence and Preemption (MLPP), will ring on the device. Also, when you enable DND, the auto answer feature gets disabled.

The user can enable and disable DND by using any of the following methods:

- Softkey
- Feature Line Key
- Cisco Unified Communications Self Care Portal

You can enable and disable DND on a per-phone basis in Cisco Unified Communications Manager Administration.

EnergyWise

The EnergyWise feature allows certain Cisco Unified IP Phones to participate in an EnergyWise-enabled system. The phone reports its power usage to the EnergyWise domain to allow the tracking and control of power within the customer premise. The phone supports alternate reduced power modes.

The following Cisco Unified IP Phones support EnergyWise in this release:

- Cisco Unified IP Phone 6901
- Cisco Unified IP Phone 6911
- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961
- Cisco Unified IP Phone 7906
- Cisco Unified IP Phone 7911
- Cisco Unified IP Phone 7931
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7945
- Cisco Unified IP Phone 7961G
- Cisco Unified IP Phone 7961G-GE
- Cisco Unified IP Phone 7962G
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971
- Cisco Unified IP Phone 7975
- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

In the Cisco Unified IP Phones, the EnergyWise feature enables the phone to sleep and wake. A sleeping phone reduces energy consumption, typically into the 0 to 1 watt range.

Limitations

You must configure the call manager to power off or power on the Cisco Unified IP Phones at least 12-13 minutes before you configure the Unified CM to power off or power on. This enables the Unified CM, switch, and Cisco Unified IP Phones to synchronize after powering on. Failure prevents the phones from powering off or entering sleep mode at the configured time.

While configuring the Unified CM, keep a minimum of 20 minutes between power off and power on. Failure prevents the phones from powering on.

EnergyWise in the Cisco Unified IP Phones 7900 Series

Cisco Unified IP Phone 7900 series phones can be configured to automatically sleep and wake at specific times. When these phones are sleeping, users cannot wake them up.

For more information about Energywise, see the appropriate user guide and administration guide:

- Cisco Unified IP Phone 7900 Series User Guide
- Cisco Unified IP Phone 7900 Series Administration Guide

EnergyWise in the Cisco Unified IP Phones 6900 8900 and 9900 Series

The Cisco Unified IP Phones 6900, 8900, and 9900 Series support EnergyWise by using configured sleep and wake times. In addition, users can wake a sleeping phone using the Select button.

For more information about Energywise, see the appropriate user guide and administration guide:

- Cisco Unified IP Phone 6901/6911 User Guide
- Cisco Unified IP Phone 6921, 6941, 6945, 6961 User Guide
- Cisco Unified IP Phone 8961, 9951, and 9971 User Guide
- Cisco Unified IP Phone 6901/6911 Administration Guide
- Cisco Unified IP Phone 6921, 6941, 6945, 6961 Administration Guide
- Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide

Hold Reversion

The Hold Reversion feature alerts a phone user when a held call exceeds a configured time limit. When the held call duration exceeds the limit, Cisco Unified Communications Manager generates alerts, such as a ring or beep, at the phone to remind the user to handle the call. The held call becomes a reverted call when the hold duration exceeds the configured time limit. For example, if you configure this feature to notify you when a call remains on hold past 30 seconds, Cisco Unified Communications Manager sends an alert, such as a ring or beep, to the phone after 30 seconds. You can also configure reminder alerts at configured intervals. A user can retrieve a reverted call on hold by going off hook, which deactivates the feature.

You configure hold reversion timers and other feature settings in Cisco Unified Communications Manager Administration for the system or for a line.

- The Hold Reversion Duration timer specifies the wait time before a reverted call alert is issued to the holding party phone.
- The Hold Reversion Notification Interval timer specifies the frequency of the periodic reminder alerts to the holding party phone.
- The Reverted Call Focus priority specifies which call type, incoming calls or reverted calls, receives focus for user actions, such as going off hook.

**Note**

SCCP phones support a minimum Hold Reversion Notification Interval (HRNI) of 5 seconds, whereas SIP phones support a minimum of 10 seconds. SCCP phones set for the minimum HRNI of 5 seconds may experience a Hold Reversion Notification ring delay of 10 seconds when handling calls involving SIP phones.

Immediate Divert

The Immediate Divert feature allows the invoker to immediately divert a call to a voice-messaging system. Managers and assistants, or anyone who shares lines, use this feature. When the call gets diverted, the line becomes available to make or receive new calls.

If the Use Legacy iDivert service parameter is set to False, the invoker can select a party voice mailbox to which to divert an incoming call. The invoker can choose between the original called party voice mailbox or the voice mailbox of the invoker.

To access the Immediate Divert feature, use the iDivert or Divert softkey. Configure the iDivert softkey by using the Softkey Template Configuration window of Cisco Unified Communications Manager Administration (the Divert softkey is not configurable; it displays automatically on the supported phone model such as Cisco Unified IP Phone 9971). The softkey template gets assigned to phones that are in the Cisco Unified Communications Manager system.

Intercom

Intercom allows a user to place a call to a predefined target. The called destination auto-answers the call in speakerphone mode with mute activated. This sets up a one-way voice path between the initiator and the destination, so the initiator can deliver a short message, regardless whether the called party is busy or idle. To ensure that the voice of the called party is not sent back to the caller when the intercom call is automatically answered, Cisco Unified Communications Manager implements whisper intercom. Whisper intercom means that only one-way audio exists from the caller to the called party. The called party must manually press a key to talk to the caller.

Internet Protocol Version 6 (IPv6)

Internet Protocol version 6 (IPv6), which is the latest version of the Internet Protocol (IP) that uses packets to exchange data, voice, and video traffic over digital networks, increases the number of network address bits from 32 bits in IPv4 to 128 bits. IPv6 support in the Cisco Unified Communications Manager network allows the network to behave transparently in a dual-stack environment and provides additional IP address space and autoconfiguration capabilities to devices that are connected to the network.

Cisco Unified IP Phones that run SIP support IPv4 only. Cisco Unified IP Phones that run SCCP can support IPv6 only, IPv4 only, or IPv4 and IPv6 in dual-stack mode.

Join

By using the Join softkey, a user can join up to 15 established calls (for a total of 16) to create a conference.

Join Across Lines

The Join Across Lines feature allows a user to join calls on multiple phone lines (either on different directory numbers or on the same directory number but on different partitions) to create a conference.

Log Out of Hunt Groups

The Log Out of Hunt Groups feature allows phone users to log their phones out from receiving calls that get routed to directory numbers that belong to line groups to which the phone lines are associated. Regardless of the phone status, the phone rings normally for incoming calls that are not calls to the line group(s) that are associated with the phone. The phone provides a visual status of the login state, so the user can determine by looking at the phone whether they are logged in to their line group(s).

The Log Out of Hunt Groups feature also comprises the following components:

- The HLog softkey allows a phone user to log a phone out of all line groups to which the phone directory numbers belong. Configure the HLog softkey in the Softkey Layout Configuration window. When the user presses the HLog softkey, the phone screen displays “Logged out of Hunt Group.” When the user presses the HLog softkey again to log back in and receive hunt group calls, the “Logged out of Hunt Group” notification on the phone screen clears.
- To enable this feature, you must configure the Hunt Group Logoff Notification service parameter, which supports the Cisco CallManager service, in the Clusterwide Parameters (Device - Phone) section of the Service Parameters Configuration window.

The Log Out of Hunt Groups feature, which is device-based, operates differently for non-shared lines than for shared lines.

Malicious Call Identification (MCID)

The MCID feature provides a useful method for tracking troublesome or threatening calls. When a user receives this type of call, the Cisco Unified Communications Manager system administrator can assign a new softkey template that adds the Malicious Call softkey to the user phone. For POTS phones that are connected to a SCCP gateway, users can use a hookflash and enter a feature code of *39 to invoke the MCID feature.

Mobile Connect and Mobile Voice Access

The Cisco Unified Mobility Mobile Connect feature enables users to manage business calls by using a single phone number and to pick up in-progress calls on the desktop phone and mobile phone. The Cisco Unified Mobility Mobile Voice Access feature extends mobile connect capabilities by way of an integrated voice response (IVR) system that is used to initiate mobile connect calls and to activate or deactivate mobile connect capabilities.

Monitoring and Recording

Cisco Unified Communications Manager supports silent call monitoring and call recording.

Call centers need to be able to guarantee the quality of customer service that an agent in a call center provides. To protect themselves from legal liability, call centers need to be able to archive agent-customer conversations.

The Silent Call Monitoring feature allows a supervisor to eavesdrop on a conversation between an agent and a customer without allowing the agent to detect the monitoring session.

The Call Recording feature allows system administrators or authorized personnel to archive conversations between the agent and the customer.

Onhook Call Transfer

The Onhook Call Transfer feature supports the onhook (hangup) action as a possible last step to complete a call transfer. You must set the Transfer On-hook Enabled service parameter, which enables onhook call transfer, to True for onhook call transfer to succeed. If the service parameter is set to False, the onhook action ends the secondary call to the third party.

In the existing implementation, if user B has an active call on a particular line (from user A) and user B has not reached the maximum number of calls on this line, the Cisco Unified IP Phone provides a Transfer softkey to user B. If user B presses the Transfer softkey (or Transfer button, if available) once, user B receives dial tone and can make a secondary call: user B dials the number of a third-party (user C). Cisco Unified Communications Manager provides a Transfer softkey to user B again. If user B presses the Transfer softkey again (or Transfer button, if available), the transfer operation completes.

With the onhook call transfer implementation, user B can hang up after dialing the number of user C, and the transfer completes. Both the existing and new implementations work in the case of a blind transfer (user B disconnects before user C answers) and also in the case of a consult transfer (user B waits for user C to answer and announces the call from user A).

The previous implementation remains unchanged: user B can press the Transfer softkey twice to complete the transfer.

Prime Line Support for Answering Calls

With prime line support for answering calls, when the phone is idle (off hook) and receives a call on any line, the primary line always gets chosen for the call. When you configure this support, going off hook makes only the first line active, even when a call rings on another line on the phone; that is, the call does not get answered on that line. In this case, the phone user must choose the other line to answer the call.

You can configure the Always Use Prime Line service parameter for the Cisco CallManager service or you can configure the Always Use Prime Line setting for devices and device profiles. The Always Use Prime Line setting displays in the following windows in Cisco Unified Communications Manager Administration.

- **System > Service Parameters (for Cisco CallManager service)**
- **Device > Phone**
- **Device > Common Phone Profile**
- **Device > Device Settings > Default Device Profile**
- **Device > Device Settings > Device Profile**

For information on how the Always Use Prime Line setting works when a phone is idle or busy, see the following table.

**Tip**

If you configure the Always Use Prime Line setting in the Service Parameter, Common Phone Profile, and in the Phone Configuration window, Cisco Unified Communications Manager uses the configuration from the Phone Configuration window.

Table 9: Always Use Prime Line Configuration

State of Phone	Configuration for Always Use Prime Line	How Feature Works
Idle	On	<p>When the phone is idle (off hook) and receives a call on any line, the primary line gets chosen for the call. Calls on other lines continue to ring, and the phone user must select those other lines to answer these calls.</p> <p>If you choose On for the Always Use Prime Line setting in the Device Profile or Default Device Profile Configuration window, a Cisco Extension Mobility user can use this feature after logging in to the device that supports Cisco Extension Mobility; that is, if you configure Cisco Extension Mobility correctly.</p>
Idle	Off	<p>When the phone is idle and receives a call on any line, the phone user answers the call from the line on which the call is received; that is, when the phone is off hook.</p>
Idle	Default	<p>If you choose Default for the Always Use Prime Line setting in the Common Phone Profile, the Device Profile, or the Default Device Profile Configuration windows, Cisco Unified Communications Manager uses the configuration from the Always Use Prime Line service parameter when determining whether a user, including a Cisco Extension Mobility user, can use this feature.</p> <p>If you choose Default for the for the Always Use Prime Line setting in the Phone Configuration window, Cisco Unified Communications Manager uses the configuration from the common phone profile.</p>
Busy	On	<p>When the phone already has a call on a line, Cisco Unified Communications Manager uses the configuration for the Maximum Number of Calls and Busy Trigger settings to determine how to route the call.</p>
Idle	On, but you also configured Auto Answer With Headset or Auto Answer with Speakerphone	<p>If you choose the Auto Answer with Headset option or Auto Answer with Speakerphone option from the Auto Answer drop-down list box in Cisco Unified Communications Manager Administration, the Auto Answer configuration overrides the configuration for the Always Use Prime Line setting.</p>

**Tip**

This feature relies on the Cisco CallManager service, so activate the service by choosing Tools > Service Activation in Cisco Unified Serviceability. In addition, you can run SDI trace for the Cisco CallManager service. When you view the log in RTMT, you can see the configured value that is used by the device; for example, alwaysPrimeLine=1, which indicates that the device uses On for the configuration.

**Note**

If you want to do so, you can configure prime line support for answering calls in the Bulk Administration Tool.

Peer-to-Peer Image Distribution (PPID)

The Peer Firmware Sharing feature provides these advantages in high-speed campus LAN settings:

- Limits congestion on TFTP transfers to centralized TFTP servers.
- Eliminates the need to manually control firmware upgrades.
- Reduces phone downtime during upgrades when large numbers of devices are reset simultaneously.

In most conditions, the Peer Firmware Sharing feature optimizes firmware upgrades in branch deployment scenarios over bandwidth-limited WAN links.

When the feature is enabled, it allows the phone to discover like phones on the subnet that are requesting the files that make up the firmware image and to automatically assemble transfer hierarchies on a per-file basis. The individual files that make up the firmware image get retrieved from the TFTP server by only the root phone in the hierarchy and are then rapidly transferred down the transfer hierarchy to the other phones on the subnet using TCP connections.

Configure PPID from the Phone Configuration window by using the Peer Firmware Sharing settings in the Product-Specific Configuration Layout. This menu option indicates whether the phone supports PPID. Settings include enabled or disabled (the default).

To configure the PPID feature for many phones, use the Peer Firmware Settings field in the Phone Template window of the Bulk Administration Tool.

For more information, see the applicable Cisco Unified IP Phone administration guide.

Quality Report Tool

The Quality Report Tool (QRT), a voice-quality and general problem-reporting tool for Cisco Unified IP Phones, allows users to easily and accurately report audio and other general problems with their IP phone. QRT gets loaded as part of the Cisco Unified Communications Manager installation, and the Cisco Extended Functions (CEF) service supports it.

As a system administrator, you enable QRT functionality by creating, configuring, and assigning a softkey template to associate the QRT softkey on a user IP phone. You can choose from two different user modes, depending upon the level of user interaction that you want with QRT. You then define how the feature will

work in your system by configuring system parameters and setting up Cisco Unified Serviceability tools. You can create, customize, and view phone problem reports by using the QRT Viewer application.

Support for the QRT feature extends to any IP phone that includes the following capabilities:

- Support for softkey templates
- Support for IP phone services
- Controllable by CTI
- Contains an internal HTTP server

When users experience problems with their IP phones, they can report the type of problem and other relevant statistics by pressing the QRT softkey on the Cisco Unified IP Phone during one of the following call states:

- Connected
- Connected Conference
- Connected Transfer
- On Hook

From a supported call state, and using the appropriate problem classification category, a user can then choose the reason code that best describes the problem that is being reported for the IP phone. A customized phone problem report provides you with the specific information.

Secure Tone

You can configure a phone to play a 2-second tone that notifies the user that a call is encrypted and that both phones on the call are configured as “protected” devices. The tone plays for both parties when the call is answered. The tone does not play unless both phones are “protected” and the call occurs over encrypted media.

Several configuration requirements exist for the secure tone to play.

Service URL

You can configure a Cisco Unified IP Phone Service URL, such as the extension mobility service, to a phone button. When the button gets pressed, the service gets invoked.

To configure a service URL on a phone button for the user, the administrator performs the following steps:

- 1 Using IP Phone Services Configuration, create a service.
- 2 Using Phone Button Configuration, create a custom phone button template to include the service URL feature.
- 3 Using Phone Configuration, add the custom phone button template to each phone that requires the service URL button.
- 4 Using Phone Configuration, subscribe to each appropriate service.
- 5 Using Phone Configuration, add the service URL button.
- 6 Notify the users to configure services for their phone by using the Add/Update your Service URL Buttons link on the User Options Menu.

Single Button Barge/cBarge

The Single Button Barge/cBarge and Privacy features work together. These features work by using only shared lines.

The Barge and cBarge features add a user to a call that is in progress. The Single Button Barge/cBarge feature allows a user to simply press the shared-line button of a call to automatically add that user to the call. The users that are currently on the call receive a tone.

Privacy allows a user to allow or disallow other users of shared-line devices to view the device call information or to allow another user to barge in to its active calls.

Speed Dial and Abbreviated Dial

Cisco Unified Communications Manager supports the configuration of up to 199 speed-dial entries, which are accessed through phone buttons and abbreviated dialing.

The administrator configures speed-dial entries and abbreviated dial indexes in the same window. From the Phone Configuration window, choose **Add/Update Speed Dials** from the Related Links drop-down list box at the top of the window and click **Go**. The Speed Dial and Abbreviated Dial Configuration window displays for this phone.

When the user configures speed-dial entries, part of the speed-dial entries can get assigned to the speed-dial buttons on the IP phone; the remaining speed-dial entries get used for abbreviated dialing. When a user starts dialing digits, the AbbrDial softkey displays, and the user can access any speed-dial entry by entering the appropriate index (code) for abbreviated dialing.

When users configure speed-dial in Cisco Unified Communications Self Care Portal, 199 entries display. Depending on the phone type, up to a maximum of 107 speed-dials can be used. Speed dials for which there is no corresponding button on the phone can only be accessed by using the Abbreviated Dial feature, if available.

Table 10: Maximum Speed Dials per Phone Model

Phone Model	Maximum Number of Speed-Dial Entries Available on the Phone	Maximum Number of Speed-Dial Entries Available with Expansion Modules
Cisco Unified IP Phone 9971	4	107
Cisco Unified IP Phone 9951	3	71
Cisco Unified IP Phone 8961	3	35
Cisco Unified IP Phone 7975	6	55
Cisco Unified IP Phone 7965, 7962	4	53
Cisco Unified IP Phone 7960	4	35

**Note**

Maximum number of speed-dial entries available on the phone is equal to maximum number of buttons available on the phone minus one button for Line 1.

VPN Client

The VPN Client feature establishes a virtual private network (VPN) connection on your phone using the Secure Sockets Layer (SSL). The VPN connection is used for situations in which a phone is located outside a trusted network or when network traffic between the phone and Cisco Unified Communications Manager must cross untrusted networks.

After the phone gets configured with VPN functionality and the VPN feature gets enabled, the user enters credentials as follows:

- If the phone is located outside the corporate network-The user is prompted at login to enter the credentials based on the authentication method that the system administrator configured on the phone.
- If the phone is located inside the corporate network:
 - If Auto Network Detection is disabled, the user is prompted for credentials, and a VPN connection is possible.
 - If Auto Network Detection is enabled, the user cannot connect through VPN so there is no prompt.

The user can enable or disable the VPN Client mode on the phone.

You can use Cisco Unified Reporting to determine which Cisco Unified IP Phones support the VPN client. From Cisco Unified Reporting, click Unified CM Phone Feature List. For the Feature, choose Virtual Private Network Client from the pull-down menu. The system displays a list of products that support the feature.

Whisper Coaching

Silent call monitoring is a feature that allows a supervisor to discreetly listen to a conversation between an agent and a customer without allowing the agent to detect the monitoring session. Whisper coaching is an enhancement to silent call monitoring feature that allows supervisors to talk to agents during a monitoring session. This feature provides applications the ability to change the current monitoring mode of a monitoring call from Silent Monitoring to Whisper Coaching and vice versa.

To invoke whisper coaching, choose On from the built-in bridge drop-down list (**Device > Phone**).

Phone Association

Users can control some devices, such as phones. Applications that are identified as users control other devices, such as CTI ports. When users have control of a phone, they can control certain settings for that phone, such as speed dial and call forwarding.

Phone Administration Tips

The following sections contain information that may help you configure phones in Cisco Unified Communications Manager Administration.

Phone Search

The following sections describe how to modify your search to locate a phone. If you have thousands of Cisco Unified IP Phones in your network, you may need to limit your search to find the phone that you want. If you are unable to locate a phone, you may need to expand your search to include more phones.

**Note**

Be aware that the phone search is not case sensitive.

Searching by Device Name

When you enter the MAC address of the device in the MAC Address field when you are adding the phone, you can search by using that value as the Device Name in the Find and List Phones window.

Searching by Description

If you enter a user name and/or extension in the Description field when you are adding the phone, you can search by using that value in the Find and List Phones window.

Searching by Directory Number

To search for a phone by its directory number (DN), choose Directory Number. Choose a search criterion (such as begins with or ends with) and either choose a directory number from the drop-down list box below the **Find** button or enter a search string. Click the **Find** button to perform the search.

**Note**

Some directory numbers do not associate with phones. To search for those directory numbers, which are called unassigned DN, use the Route Plan Report window or use the Directory Number Configuration Find/List window.

Searching by Calling Search Space

If you choose calling search space, the options that are available in the database display; you can choose one of these options from the drop-down list box below the **Find** button.

Searching by Device Pool

If you choose device pool, the options that are available in the database display (for example, Default); you can choose one of these options from the drop-down list box below the **Find** button.

Searching by Device Type

To search for a phone by its device type, choose Device Type and either enter a device type or choose a device type from the drop-down list box below the **Find** button.

Searching by Call Pickup Group

To search for a phone by its call pickup group, choose Call Pickup Group. If you choose Call Pickup Group, the options that are available in the database display; you can choose one of these options from the drop-down list box below the **Find** button. Alternatively, click the **Find** button only.

Searching by LSC Status

If you choose LSC status, the options that are available in the database display (for example, Operation Pending); you can choose one of these options from the drop-down list box below the **Find** button.

Searching by Authentication String

To search for a phone by an authentication string, choose Authentication String and enter an authentication string.

Searching by Device Protocol

To search for a phone by the protocol, choose Device Protocol and either enter a protocol, such as SIP, or choose a protocol from the drop-down list box below the **Find** button.

Searching by Security Profile

To search for a phone by its security profile, choose Security Profile and either enter a security profile name or choose a security profile from the drop-down list box below the **Find** button.

Searching by Common Device Configuration

To search for a phone by its common device configuration, choose Common Device Configuration and either enter a common device configuration name or choose a common device configuration from the drop-down list box below the **Find** button.

Refining Search Criteria

To add additional search criteria, click the + button. When you add criteria, the system searches for a record that matches all criteria that you specify. To remove criteria, click the - button to remove the last added criterion or click the **Clear Filter** button to remove all added search criteria.

Finding All Phones in the Database

To find all phones that are registered in the database, choose Device Name from the list of fields; choose "is not empty" from the list of patterns; then, click the **Find** button.

**Note**

The list in the Find and List Phones window does not include analog phones and fax machines that are connected to gateways (such as a Cisco VG200). This list shows only phones that are configured in Cisco Unified Communications Manager Administration.

Messages Button

By performing the following actions, you can configure a voice-messaging access number for the messages button on Cisco Unified IP Phone, so users can access the voice-messaging system by simply pressing the messages button:

- 1 Configure the voice-mail pilot number by choosing **Advanced Features > Voice Mail > Voice Mail Pilot**.
- 2 Configure the voice-mail profile by choosing **Advanced Features > Voice Mail > Voice Mail Profile**.
- 3 Choose the appropriate profile from the Voice Mail Profile field on the Directory Number Configuration window. By default, this field uses the default voice-mail profile that uses the default voice-mail pilot number configuration.



Note

Typically, you can edit the default voice-mail pilot and default voice-mail profiles to configure voice-messaging service for your site.

Directories Button

The Cisco Unified IP Phone can display directories of names and phone numbers. You access this directory from the directories button on the IP phone. For end users to retrieve contacts from the corporate directory, the administrator must enter users into the directory. Enter the contacts one at a time by using Cisco Unified Communications Manager Administration User Management (**User Management > End User**). The administrator can also add multiple users in bulk by using the Bulk Administration Tool (**Bulk Administration > End User**).

Other types of directories exist that can display on the IP phone: personal directory and phone directory (such as missed calls). To find out about these directories, see the user guide for the specific Cisco Unified IP Phone.

The URL Directories enterprise parameter defines the URL that points to the global directory for display on the Cisco Unified IP Phone. The XML device configuration file for the phone stores this URL.



Tip

If you are using IP addresses rather than DNS for name resolution, make sure that the URL Directories enterprise parameter value uses the IP address of the server for the hostname.



Tip

If the phone URL was not updated correctly after the URL Directories enterprise parameter was changed, try stopping and restarting the Cisco TFTP service; then, reset the phone.

Cisco Unified Communications Self Care Portal

Cisco Unified IP Phone users access Cisco Unified Communications Self Care Portal through their web browser, so they can configure a variety of features on their phone. Some of the configurable features include user locale, user password, call forward, speed dial, and remote destinations. By setting enterprise parameters

as either True or False, you can configure which features are made available to users; for example, you can set the Show Speed Dial Settings enterprise parameter to False, and users cannot configure speed dials on their phones.

Maximum Phone Fallback Queue Depth Service Parameter

The Cisco CallManager service uses the Maximum Phone Fallback Queue Depth service parameter to control the number of phones to queue on the higher priority Cisco Unified Communications Manager when that Cisco Unified Communications Manager is available for registration. The default specifies 10 phones per second. If a primary Cisco Unified Communications Manager fails, the phones fail over to the secondary Cisco Unified Communications Manager. The failover process happens as fast as possible by using the priority queues to regulate the number of devices that are currently registering.

When the primary Cisco Unified Communications Manager recovers, the phones get returned to that Cisco Unified Communications Manager; however, you do not need to remove a phone from a working Cisco Unified Communications Manager, in this case the secondary system, as fast as possible because the phone remains on a working system. The queue depth gets monitored (using the Maximum Phone Fallback Queue Depth service parameter setting) to determine whether the phone that is requesting registration gets registered now or later. If the queue depth is greater than 10 (default), the phone stays where it is and tries later to register to the primary Cisco Unified Communications Manager.

In the Service Parameters Configuration window, you can modify the Maximum Phone Fallback Queue Depth service parameter. If the performance value is set too high (the maximum setting specifies 500), phone registrations could slow the Cisco Unified Communications Manager real-time response. If the value is set too low (the minimum setting specifies 1), the total time for a large group of phones to return to the primary Cisco Unified Communications Manager will be long.

Dependency Records

If you need to find what directory numbers a specific phone is using or to what phones a directory number is assigned, choose Dependency Records from the Related Links drop-down list box on the Cisco Unified Communications Manager Administration Phone Configuration or Directory Number Configuration window. The Dependency Records Summary window displays information about directory numbers that are using the phone. To find more information about the directory number, click the directory number, and the Dependency Records Details window displays. If the dependency records are not enabled for the system, the dependency records summary window displays a message.

Phone Failover And Fallback

This section describes how phones fail over and fall back if the Cisco Unified Communications Manager to which they are registered becomes unreachable. This section also covers conditions that can affect calls that are associated with a phone, such as reset or restart.

Cisco Unified Communications Manager Fails or Becomes Unreachable

The active Cisco Unified Communications Manager designation applies to the Cisco Unified Communications Manager from which the phone receives call-processing services. The active Cisco Unified Communications Manager usually serves as the primary Cisco Unified Communications Manager for that phone (unless the primary machine is not available).

If the active Cisco Unified Communications Manager fails or becomes unreachable, the phone attempts to register with the next available Cisco Unified Communications Manager in the Cisco Unified Communications Manager Group that is specified for the device pool to which the phone belongs.

The phone device reregisters with the primary Cisco Unified Communications Manager as soon as it becomes available after a failure. See the [Maximum Phone Fallback Queue Depth Service Parameter](#), on page 89 for information about phone registration during failover.

When using an IP phone's VPN feature and the phone's VPN connection must failover between VPN capable devices, it will take eight minutes for the phone to failover. The phone will try to reconnect to the primary VPN 15 times before failing over to the next VPN connection.

**Note**

Phones do not fail over or fall back while a call is in progress.

Phone is Reset

If a call is in progress, the phone does not reset until the call finishes.