CHAPTER 44

Cisco Unified IP Phones

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.

This section covers the following topics:

Supported Cisco Unified IP Phones

Table 44-1 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 7900 family (SCCP and SIP)
- Cisco Unified IP Video Phone 7985
- Cisco Unified IP Phone Expansion Module 7914, 7915, and 7916
- Cisco IP Conference Station 7935 and 7936
- Cisco IP Phone 30 VIP
- Cisco IP Phone 12 series

For the latest information on features and services that these phone models support, refer to 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

Table 44-1 Supported Cisco Unified IP Phones and Features

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7970 and Cisco Unified IP Phone 7971	The Cisco Unified IP Phone 7970 and 7971, full-featured, eight-line business sets, support SCCP and SIP and the following features: • A backlit, color touchscreen display for easy access to call details and features. • Four fixed feature buttons: – Messages—Accessing voice-messaging messages – Settings—Adjusting phone settings – Services—Accessing services – Directories—Accessing call logs and directories • A Help (?) button for immediate assistance with calling features • Eight programmable buttons to use as line buttons, speed-dial buttons, or for other phone services • Five softkeys for accessing additional call details and functionality (Softkeys change depending on the call state for a total of 16 softkeys.) • An internal, two-way, full-duplex speakerphone and microphone mute The Cisco Unified IP Phone 7970/71G-GE represent the gigabit ethernet version of the Cisco Unified IP Phone 7970/71 while Cisco Unified IP Phone 7970G represents the non-gigabit version.

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7960 and Cisco Unified IP Phone 7961	The Cisco Unified IP Phone 7960 and 7961, full-featured, six-line business sets, support SCCP and SIP and the following features: • 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 The Cisco Unified IP Phone 7961G-GE represents the gigabit ethernet version of the Cisco Unified IP Phone 7961 while Cisco Unified IP Phone 7961G represents the non-gigabit version.
Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7941	The Cisco Unified IP Phone 7940 and 7941, two-line business sets with features similar to the Cisco Unified IP Phone 7960, support SCCP and SIP and include the following features: • 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 The Cisco Unified IP Phone 7941G-GE represents the gigabit ethernet version of the Cisco Unified IP Phone 7941 while Cisco Unified IP Phone 7941G represents the non-gigabit version.

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7931	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 • 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 Wireless IP Phone 7920	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 • 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

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone Expansion Module 7914	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. **Note** You can create the Cisco Unified IP Phone Expansion Module 7914 phone button template by renaming the phone button template that is used for the standard Cisco Unified IP Phone 7960. Refer to "Phone Button Template Configuration" in the *Cisco Unified Communications Manager Administration Guide* for more information. The Cisco Unified IP Phone Expansion Module 7914 includes an LCD to identify the function of the button and the line status. You can daisy chain two Cisco Unified IP Phone Expansion Modules 7914 to provide 28 additional lines or speed-dial and feature buttons.
Cisco Unified IP Phone Expansion Module 7915 and Cisco Unified IP Phone Expansion Module 7916	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. **Note** You can create the Cisco Unified IP Phone Expansion Module phone button template by renaming the phone button template that is used for the standard Cisco Unified IP Phone 7960. Refer to "Phone Button Template Configuration" in the *Cisco Unified Communications Manager Administration Guide* for more information. The Cisco Unified IP Phone Expansion Module 7915 and 7916 includes an LCD to identify the function of the button and the line status. 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.
Cisco Unified IP Phone 7912	The Cisco Unified IP Phone 7912, which is a single-line phone that supports a maximum of two calls at the same time, supports SCCP and SIP and provides basic-feature functionality for individuals who conduct low to medium telephone traffic. This phone, which supports inline power, provides an integrated 10/100 Ethernet switch for connectivity to a collocated PC. This phone offers four dynamic softkeys.

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7911	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. Similarities exist between the Cisco Unified IP Phone 7911 menus and the Cisco Unified IP Phone 7970 menus. The Applications Menu button opens up a main applications menu. This phone, which supports inline power, provides an integrated 10/100 Ethernet switch for connectivity to a collocated PC. This phone offers four dynamic softkeys.
Cisco Unified IP Phone 7910	The Cisco Unified IP Phone 7910, a single-line, basic-feature phone that is designed primarily for common-use areas with medium telephone traffic such as lobbies or breakrooms, includes the following features: • Four dedicated feature buttons for Line, Hold, Transfer, and Settings • Six programmable feature buttons that you can configure through phone button templates in Cisco Unified Communications Manager Available features include Call Park, Redial, Speed Dial, Call Pickup, Conference, Forward All, Message Waiting, and Meet-Me Conference. • A two-line LCD (24 characters per line) that indicates the directory number, call status, date, and time • An internal speaker that is designed for hands-free dialing.
Cisco Unified IP Phone 7906	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. Similarities exist between the Cisco Unified IP Phone 7906 menus and the Cisco Unified IP Phone 7970 menus. The Applications Menu button opens up a main applications menu. This phone, which supports inline power, provides an integrated 10/100 Ethernet switch for connectivity to a collocated PC. This phone offers four dynamic softkeys.

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco Unified IP Phone 7905	The Cisco Unified IP Phone 7905, a low-cost, single-line, basic-feature phone that is designed primarily for common-use areas such as cafeterias, break rooms, lobbies, and manufacturing floors, supports SCCP and SIP and includes the following features: • LCD that displays features such as time, date, phone number, caller ID, call status, and softkey tabs • Four softkeys that engage the function that displays on the corresponding LCD tabs. (Softkey functions change depending on the status of the phone.) • Three dedicated buttons for Hold, Menu, and Navigation • An internal speaker that is designed for hands-free dialing
Cisco Unified IP Phone 7902	The Cisco Unified IP Phone 7902 provides a cost-effective, entry-level IP phone for a lobby, laboratory, manufacturing floor, or another area where only basic calling capability is required. The single-line Cisco Unified IP Phone 7902 includes the following features: • Fixed feature keys that provide one-touch access to the redial, transfer, conference, and voice-messaging access features • Three dedicated buttons for hold, menu, and volume control • Inline power that allows the phone to receive power over the LAN
Cisco Unified IP Phone 7985	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: • Color screen • Support for up to eight line or speed-dial numbers • Context-sensitive online help for buttons and feature

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco Unified IP Conference Station 7936	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: • 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 IP Conference Station 7935	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: • 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 IP Phone 12 SP+	The Cisco IP Phone 12 SP+ offers many of the same features as PBX or POTS telephones. This IP phone includes the following features: • 12 programmable line and feature buttons • An LED that is associated with each of the 12 feature and line buttons to indicate feature and line status • A two-line LCD (20 characters per line) for call status and identification • An internal, two-way speakerphone and microphone mute

Table 44-1 Supported Cisco Unified IP Phones and Features (continued)

Cisco Unified IP Phone Model	Description
Cisco IP Phone 30 VIP	The Cisco IP Phone 30 VIP offers many of the same features as PBX or POTS telephones. This IP phone includes the following features: • 26 programmable line and feature buttons • An LED that is associated with each of the 26 feature and line buttons to indicate feature and line status • A two-line LCD for displaying date and time, calling party name, calling party number, and digits dialed • An internal, two-way speakerphone with microphone mute • Dedicated feature buttons for Transfer, Hold, and Redial
Cisco Unified SIP Phone 3951	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.

Cisco Unified IP Phones That Support SIP

Cisco Unified Communications Manager supports SIP on the following Cisco Unified IP Phones:

- Cisco Unified IP Phone 7975
- Cisco Unified IP Phone 7970/71
- Cisco Unified IP Phone 7965/45
- Cisco Unified IP Phone 7960/61
- Cisco Unified IP Phone 7940/41
- Cisco Unified IP Phone 7911/06
- Cisco Unified IP Phone 7905/12

The administrator uses the Cisco Unified Communications Manager Administration Phone Configuration window to configure an IP phone for SCCP or SIP. If SIP is chosen, additional Cisco Unified Communications Manager Administration configuration windows get used to configure SIP; for example, SIP Profile Configuration. See Table 44-8 for configuration requirements. For information on SIP profiles and SIP dial rules, see "SIP Dial Rules Configuration" and "SIP Profile Configuration" sections in the *Cisco Unified Communications Manager Administration Guide.*

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" section on page 18-2.

For instructions on how to configure H.323 clients and CTI ports, refer to "Cisco Unified IP Phone Configuration" section in the *Cisco Unified Communications Manager Administration Guide*.

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 7970 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 Administrator's 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.

For details of Cisco Unified Mobile Communicator, including configuration to integrate with Cisco Unified Mobility capabilities and features, refer to the "Cisco Unified Mobile Communicator" section in the *Cisco Unified Communications Manager Features and Services Guide*.

Codec Usage

Release 5.1(3) of Cisco Unified Communications Manager and subsequent releases support the Advertise G.722 Codec enterprise parameter, which determines whether Cisco Unified IP Phones will 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. This parameter only applies to Cisco Unified IP Phone 7941G, 7941G-GE, 7961G, 7961G-GE, 7970G, and 7971G-GE. 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 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.

Table 44-2 indicates how the phone responds to the configuration options.

Table 44-2 How Phone Responds to Configuration Settings

Enterprise Parameter Setting	Phone (Product-Specific) Parameter Setting	Phone Advertises G.722
Advertise G.722 Codec Enabled (True)	Use System Default	Yes
Advertise G.722 Codec Enabled (True)	Enabled	Yes
Advertise G.722 Codec Enabled (True)	Disabled	No
Advertise G.722 Codec Disabled (False)	Use System Default	No
Advertise G.722 Codec Disabled (False)	Enabled	Yes
Advertise G.722 Codec Disabled (False)	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 7971G-GE
- Cisco Unified IP Phone 7970G
- Cisco Unified IP Phone 7961G-GE
- Cisco Unified IP Phone 7961G
- Cisco Unified IP Phone 7941G-GE
- Cisco Unified IP Phone 7941G

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 The standard phone button template for the Cisco Unified IP Phone 7960, which supports the Cisco Unified IP Phone Expansion Module 7914, includes buttons for both devices (up to 34 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. Refer to 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 7900 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. Table 44-3 provides descriptions of the standard phone button templates.

Table 44-3 Default Phone Button Templates Listed by Model

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.
Standard 7960 SCCP	The Standard 7960 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 7960.

Table 44-3 Default Phone Button Templates Listed by Model (continued)

Phone Button Template Name	Template Description
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	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	The Standard 7931 SCCP template uses 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.
Standard 7912 SCCP	The Standard 1912 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	The Standard 7911 SCCP 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 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.

Table 44-3 Default Phone Button Templates Listed by Model (continued)

Phone Button Template Name	Template Description
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	The Standard 7906 SCCP 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 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.
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+	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. **Note** For only the Cisco IP Phone 30 SP+, assign button 26 for automatic echo cancellation (AEC).
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.

Table 44-3 Default Phone Button Templates Listed by Model (continued)

Phone Button Template Name	Template Description
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.
Default VGC Virtual Phone	The Default VGC Virtual Phone template for the Cisco VGC Virtual Phone 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).
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 Analog	The Standard Analog template for analog phones uses button 1 for line 1.
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.
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.

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 7970/71, 7960/61, 7940/41, 7911, 7906—Line (one or more)
  - Cisco Unified IP Phone 7931—Line
  - Cisco Unified IP Phone 7912—Line, speed dial, hold, and settings

  - Cisco Unified IP Phone 7910—Forward all
  - Cisco Unified IP Phone 7905 and 7902—Line, speed dial, hold, and settings
  - Cisco Wireless IP Phone 7920—Line (one or more)
  - Cisco IP Phone 12 SP+—Line (one or more), hold, call park, and forward all
  - Cisco IP Phone 30 VIP—Line (one or more), call park, and forward all
  - 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 44-4.

Table 44-4 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.
Call Park BLF	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. **Note** Use this button with the directed call park feature (a transfer function), not with the standard call park feature (a hold function).

Table 44-4 Phone Button Feature Description (continued)

Feature	Description
Conference	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.) 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. Refer to the "Conference Bridges" chapter for more information.
Forward all	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.
Hold	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.
Line	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.
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.
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.
Privacy	Users press this button to activate/deactivate privacy.
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 CM User Options menu.
BLF/SpeedDial	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 to the right of the display), 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 44-5). 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). See "Configuring Phone Button Templates" and "Modifying Phone Button Template Button Items" sections in the *Cisco Unified Communications Manager Administration Guide*.

Table 44-5 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
HLog (Hunt Group)	Yes	Yes
Mobility	Yes	Yes
Settings	No, uses existing button	Yes
Information	No, uses existing button	No

Table 44-5 Programmable Line Keys for Cisco Unified IP Phones (continued)

Feature	Phone Model 7971, 7970, 7961, 7941, 7914, 7915, 7916	Phone Model 7931 (SCCP only)
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 Cisco Call Back on the Cisco Unified IP Phones. The administrator uses the Softkey Template Configuration windows in Cisco Unified Communications Manager Administration to create and update 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 Feature
- Standard Assistant
- Standard Protected Phone
- Standard Shared Mode Manager
- Standard Manager

Note The default process does not assign a softkey template to the Cisco Unified IP Phone. The administrator 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.

The administrator creates 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)
- User Device Profile Configuration
- Default Device Profile Configuration

Add Application

The administrator 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 the administrator 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 Delete Application button allows the administrator 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**, Softkey Layout Configuration displays.

The Softkey Layout Configuration pane contains the following fields:

- Call states—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 44-6 lists the call states.

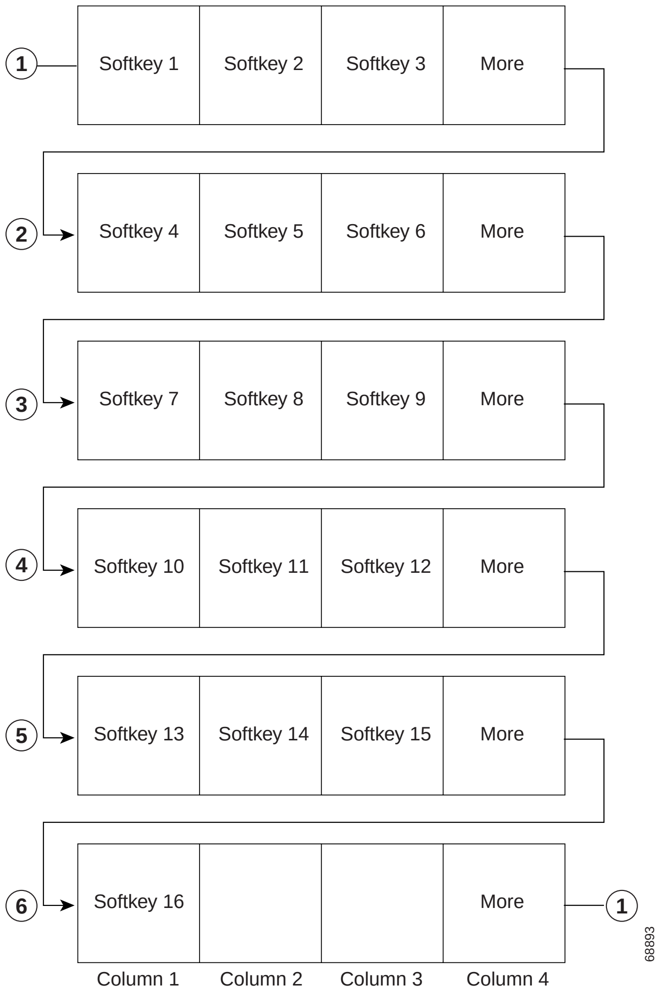
Table 44-6 Call States

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 Figure 44-1 for a sample softkey layout.

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.

Figure 44-1 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 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. See "Softkey Template Configuration" in the *Cisco Unified Communications Manager Administration Guide*.

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, however, modify nor delete the Standard Common Phone Profile.

For information on configuring common phone profiles, refer to the "Common Phone Profile Configuration" in the *Cisco Unified Communications Manager Administration Guide*.

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. For information on enabling autoregistration, refer to "Enabling Autoregistration" in the *Cisco Unified Communications Manager Administration Guide*. 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, as described in the *Cisco Unified Communications Manager Security Guide.*

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. Refer to the *Cisco Unified Communications Manager Bulk Administration Guide* for detailed instructions on using BAT.

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. See "User/Phone Add Configuration" in the *Cisco Unified Communications Manager Administration Guide* for configuration details.

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 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.

For information about features that are related to directory numbers, see the "Directory Number Features" section on page 18-6. The following features get configured for directory numbers: call forward and call waiting.

See the following sections for supported phone features:

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.

For more information about Barge, and Privacy, refer to Barge and Privacy in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

For information on how to configure this feature for your phone, refer to the "Calling Party Normalization" chapter in the *Cisco Unified Communications Manager Features and Services Guide*.

You can configure the international escape character, +, to globalize the calling party number. For information on the international escape character, +, see the "Using the International Escape Character +" section in the *Cisco Unified Communications Manager System Guide*.

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.

The following types of call forward exist:

- Call Forward All, including CFA Destination Override, CFA Loop Prevention, and CFA Loop Breakout, page 44-30
- Call Forward Busy, page 44-31
- Call Forward No Answer, page 44-31
- Call Forward No Coverage, page 44-32

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. See Service Parameters Configuration in the *Cisco Unified Communications Manager Administration Guide* for information about configuring service parameters.

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 CFA 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 CM User Options windows, Cisco Unified Communications Manager does not prevent the CFA loop.

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 in a cluster 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.

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 in a cluster, 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 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 the Cisco Unified IP Phones 7911, 7941, 7961, 7970, and 7971. Support includes end-of-call reporting, mid-call reporting (for example, call hold, media disconnect), and voice quality metrics. Cisco Unified IP Phones 7905, 7912, 7940, and 7960 that are running SIP do not report voice quality metrics or mid-call 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.

For information on configuring the Cisco Unified IP Phones for which call diagnostics and voice-quality metrics are available, refer to the Cisco Unified IP Phone user and administration documentation.

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. For more information about call park, refer to "Call Park and Directed Call Park" in the *Cisco Unified Communications Manager Features and Services Guide*.

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). For more information about call pickup, refer to "Call Pickup" in the *Cisco Unified Communications Manager Features and Services Guide*.

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. For information about configuring call pickup notification, refer to "Call Pickup" in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

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.

For more information about connected number display, see "Call Display Restrictions" in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

For information about configuring Device Mobility, see "Device Mobility" in the *Cisco Unified Communications Manager Features and Services Guide*.

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 "Making and Receiving Multiple Calls Per Directory Number" section on page 18-7.

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. Configured directed call park numbers exist cluster wide. 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.

For more information about directed call park, refer to "Call Park and Directed Call Park" in the *Cisco Unified Communications Manager Features and Services Guide*.

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 CM User Options windows

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

For more information on the Do Not Disturb feature, see "Do Not Disturb" in the *Cisco Unified Communications Manager Features and Services 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.

As administrator, you configure hold reversion timers and other feature settings in Cisco Unified Communications Manager Administration for the cluster 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.

For more information about hold reversion, see "Hold Reversion" in the *Cisco Unified Communications Manager Features and Services Guide*.

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 softkey. Configure this softkey by using the Softkey Template Configuration window of Cisco Unified Communications Manager Administration. The softkey template gets assigned to phones that are in the Cisco Unified Communications Manager system.

For more information about Immediate Divert, refer to "Immediate Divert" in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

For information on configuring Intercom, see the *Cisco Unified Communications Manager Administration Guide* and the Intercom chapter in the *Cisco Unified Communications Manager Features and Services Guide*.

Join

By using the Join softkey, a user can join up to 15 established calls (for a total of 16) to create a conference. For more information about Join, see the "Making and Receiving Multiple Calls Per Directory Number" section on page 18-7.

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. For more information about the Join Across Lines feature, see the "Making and Receiving Multiple Calls Per Directory Number" section on page 18-7.

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.

For more information about MCID, refer to the "Malicious Call Identification" chapter in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

For more information about mobile connect and mobile voice access, see "Cisco Unified Mobility" in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

For more information about monitoring and recording, see "Monitoring and Recording" in the *Cisco Unified Communications Manager Features and Services Guide*.

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.

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. See the *Cisco Unified Communications Manager Bulk Administration Guide*.

For more information, refer to 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 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

Note For more information, refer to the following URL for the appropriate Cisco Unified IP Phone guide for your phone:
http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html.

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.

For detailed information about configuring and using the Quality Report Tool feature, refer to "Quality Report Tool" in the *Cisco Unified Communications Manager Features and Services Guide.* For more information about configuring and using the QRT Viewer, refer to the *Cisco Unified Serviceability Administration Guide.*

For information about the user interface, refer to the appropriate Cisco Unified IP Phone Guide for your IP phone and the *Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager*.

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. For a detailed description of the secure-tone feature and the configuration requirements, see the *Cisco Unified Communications Manager Security Guide*.

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 Cisco Unified 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.

For more information about the Single Button Barge/cBarge feature, refer to "Barge and Privacy" in the *Cisco Unified Communications Manager Features and Services Guide*.

Speed Dial and Abbreviated Dial

Cisco Unified Communications Manager supports the configuration of up to 99 speed-dial entries, which are accessed through phone buttons and abbreviated dialing.

When the user configures up to 99 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. For information about configuring speed dials, see "Configuring Speed-Dial Buttons" in the *Cisco Unified Communications Manager Administration Guide*.

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. For more information on associating phones with users, refer to the "Associating Devices to an End User" section in the *Cisco Unified Communications Manager Administration Guide*.

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. For information on finding devices that have users actively logged in to them, see "Finding an Actively Logged-In Device", *Cisco Unified Communications Manager Administration Guide*.

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 7970, 7960, and 7940, so users can access the voice-messaging system by simply pressing the messages button:

1. Configure the voice-mail pilot number by choosing **Voice Mail > Voice Mail Pilot**.
2. Configure the voice-mail profile by choosing **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.

For more information on configuring a voice-messaging service, refer to the "Voice Mail Connectivity to Cisco Unified Communications Manager" chapter.

Note For the Cisco IP Phone 12 SP+ and 30 VIP, you can use phone button templates to configure a button with the message-waiting feature for access to a voice-messaging service.

Directories Button

The Cisco Unified IP Phone 7970, 7960, and 7940 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, refer to 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 Cisco Unified IP Phone 7970, 7960, and 7940. 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.

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 CM User Options

Cisco Unified IP Phone users access Cisco Unified CM User Options 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 personal address book. By setting enterprise parameters as either True or False, administrators can configure which features are made available to users; for example, the administrator can set the Show Speed Dial Settings enterprise parameter to False, and users cannot configure speed dials on their phones.

For more information on how to access and use Cisco Unified CM User Options, refer to the phone guide for the specific Cisco Unified IP Phone.

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.

For more information about Dependency Records, refer to the "Accessing Dependency Records" section and the "Removing a Directory Number from a Phone" section in the *Cisco Unified Communications Manager Administration Guide*.

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" section on page 44-44 for information about phone registration during failover.

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.

Phone Configuration Checklist

Table 44-7 provides steps to manually configure phone that runs SCCP in Cisco Unified Communications Manager Administration. If you are using autoregistration, Cisco Unified Communications Manager adds the phone and automatically assigns the directory number.

Table 44-8 provides steps 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.

Table 44-7 Phone Configuration Checklist for SCCP

Configuration Steps		Procedures and Related Topics
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, page 44-41
Step 2	Add and configure the phone.	Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
Step 3	Add and configure lines (DNs) on the phone. You can also configure phone features such as call park, call forward, and call pickup.	Configuring a Directory Number, *Cisco Unified Communications Manager Administration Guide*
Step 4	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 CM User Options.	Configuring Speed-Dial Buttons, *Cisco Unified Communications Manager Administration Guide*
Step 5	Configure Cisco Unified IP Phone services. You can configure services for Cisco Unified IP Phone 7970/71, 7960/61, 7940/41, 7912, and 7905 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 CM User Options.	Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
Step 6	Customize phone button templates and softkey templates, if required. Configure templates for each phone.	Configuring Phone Button Templates, *Cisco Unified Communications Manager Administration Guide* Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide* Creating Nonstandard Softkey Templates, *Cisco Unified Communications Manager Administration Guide*
Step 7	Configure the Busy Lamp Field feature, if required. You must use customized phone button templates to configure BLF/SpeedDial buttons.	BLF/Speed Dial Configuration Settings, *Cisco Unified Communications Manager Administration Guide*

Table 44-7 Phone Configuration Checklist for SCCP (continued)

Configuration Steps		Procedures and Related Topics
Step 8	Assign services to phone buttons, if required.	Adding an IP Phone Service to a Phone Button, *Cisco Unified Communications Manager Administration Guide*
Step 9	Provide power, install, verify network connectivity, and configure network settings for the Cisco Unified IP Phone.	*Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager*
Step 10	Associate user with the phone (if required).	Configuring User-Related Information for End Users, *Cisco Unified Communications Manager Administration Guide*
Step 11	Make calls with the Cisco Unified IP Phone.	Refer to the user guide for your Cisco Unified IP Phone.

Table 44-8 lists the configuration steps for Cisco Unified IP Phones that support SIP. For third-party phones that run SIP, see the Configuration Checklist for Third-Party Phones That Are Running SIP in the *Cisco Unified Communications Manager Administration Guide*.

Table 44-8 Phone Configuration Checklist for SIP

Configuration Steps		Procedures and Related Topics
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, page 44-41
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.	Cisco Unified Communications Manager Configuration, *Cisco Unified Communications Manager Administration Guide*
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.	Phone Security Profile Configuration, *Cisco Unified Communications Manager Administration Guide* Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
Step 4	Configure the SIP Profile. The SIP Profile gets added to the phone that runs SIP by choosing the profile in the Phone Configuration window.	Configuring SIP Profiles, *Cisco Unified Communications Manager Administration Guide* Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*

Table 44-8 Phone Configuration Checklist for SIP (continued)

Configuration Steps		Procedures and Related Topics
Step 5	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.	Configuring the Phone NTP References, *Cisco Unified Communications Manager Administration Guide* Configuring a Date/Time Group, *Cisco Unified Communications Manager Administration Guide* Configuring a Device Pool, *Cisco Unified Communications Manager Administration Guide* Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
Step 6	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	Configuring SIP Dial Rules, *Cisco Unified Communications Manager Administration Guide* Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
Step 7	Add and configure the phone that runs SIP.	Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
Step 8	Add and configure lines (DNs) on the phone. You can also configure phone features such as call park, call forward, and call pickup.	Configuring a Directory Number, *Cisco Unified Communications Manager Administration Guide*
Step 9	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 CM User Options.	Configuring Speed-Dial Buttons, *Cisco Unified Communications Manager Administration Guide*
Step 10	Configure Cisco Unified IP Phone services. You can configure services for Cisco Unified IP Phone 7970/71, 7960/61, 7940/41, 7912, 7911, and 7905 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 CM User Options window.	Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*

Table 44-8 Phone Configuration Checklist for SIP (continued)

Configuration Steps		Procedures and Related Topics
Step 11	Customize phone button templates and softkey templates, if required. Configure templates for each phone.	Configuring Phone Button Templates, *Cisco Unified Communications Manager Administration Guide* Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide* Creating Nonstandard Softkey Templates, *Cisco Unified Communications Manager Administration Guide*
Step 12	Configure the Busy Lamp Field feature, if required. You must use customized phone button templates to configure BLF/SpeedDial buttons.	BLF/Speed Dial Configuration Settings, *Cisco Unified Communications Manager Administration Guide*
Step 13	Assign services to phone buttons, if required.	Adding an IP Phone Service to a Phone Button, *Cisco Unified Communications Manager Administration Guide*
Step 14	Provide power, install, verify network connectivity, and configure network settings for the Cisco Unified IP Phone.	*Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager*
Step 15	Associate user with the phone (if required).	Associating Devices to an End User, *Cisco Unified Communications Manager Administration Guide*
Step 16	Make calls with the Cisco Unified IP Phone.	Refer to the user guide for your Cisco Unified IP Phone.

Where to Find More Information

Related Topics

- Understanding Directory Numbers, page 18-1
- Voice Mail Connectivity to Cisco Unified Communications Manager, page 30-1
- Enabling Autoregistration, *Cisco Unified Communications Manager Administration Guide*
- Configuring Cisco Unified IP Phones, *Cisco Unified Communications Manager Administration Guide*
- Associating Devices to an End User, *Cisco Unified Communications Manager Administration Guide*
- User/Phone Add Configuration, *Cisco Unified Communications Manager Administration Guide*
- Phone Button Template Configuration, *Cisco Unified Communications Manager Administration Guide*
- Common Phone Profile Configuration, *Cisco Unified Communications Manager Administration Guide*
- Configuring SIP Dial Rules, *Cisco Unified Communications Manager Administration Guide*
- Configuring SIP Profiles, *Cisco Unified Communications Manager Administration Guide*

- Configuring the Phone NTP References, *Cisco Unified Communications Manager Administration Guide*
- Service Parameters Configuration, *Cisco Unified Communications Manager Administration Guide*
- Barge and Privacy, *Cisco Unified Communications Manager Features and Services Guide*
- Call Park and Directed Call Park, *Cisco Unified Communications Manager Features and Services Guide*
- Call Pickup, *Cisco Unified Communications Manager Features and Services Guide*
- Immediate Divert, *Cisco Unified Communications Manager Features and Services Guide*
- Quality Report Tool, *Cisco Unified Communications Manager Features and Services Guide*
- Presence, *Cisco Unified Communications Manager Features and Services Guide*
- Calling Party Normalization, *Cisco Unified Communications Manager Features and Services Guide*
- Using the International Escape Character +, *Cisco Unified Communications Manager System Guide*

Additional Cisco Unified Communications Manager Documentation

- Phone administration documentation that supports your phone and this version of Cisco Unified Communications Manager
- Cisco Unified IP Phone user documentation
- Firmware release notes for your phone
- *Cisco Unified Communications Manager Bulk Administration Guide*
- *Cisco Unified Communications Manager Security Guide*
- *Cisco Unified Communications Manager Assistant User Guide*
- *Cisco IP Communicator Administration Guide*