Cisco Unified CallManager and Cisco Unified IP Phone A - Z Feature Guide

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As a system administrator, you can use this comprehensive listing of Cisco Unified CallManager Cisco Unified IP Phone features as a quick reference to user and administrator features.

Note Phone features can vary according to phone model and Cisco Unified CallManager release. Some features listed in this guide are unavailable on some of the Cisco Unified IP Phones and some releases of Cisco Unified CallManager.

Abbreviated Dialing

Allows the phone user to quickly dial a phone number by entering an assigned index code (1-99) on the phone keypad.

Abbreviated dialing can be useful if the phone model does not provide speed-dial buttons or if the phone user wants to configure more speed-dial numbers than the number of speed-dial buttons on the phone.

The phone user can assign index codes from the User Options web pages.

Associated softkey: **AbbrDial**

See also Fast Dial and Speed Dialing.

Access Codes

Allows you to identify codes in the dial plan as requests for specific services, such as an outside line (dial 9) and system operator (dial 0).

Ad Hoc Conference - Advanced

Allows you to offer the following advanced conference services:

- Permit a conference participant other than the controller to add or remove participants.

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- Permit conference participants to chain multiple ad hoc conferences together in linear or non-linear fashion.
- Enable or disable advanced capabilities by defining service parameters.

Alarms

Allows you to receive important information regarding system events. You can configure the reported alarm levels and the monitors that receive alarms (such as syslog and SDI traces).

Alerts

Allows you to monitor the status of system activities.

See also Real Time Monitoring Tool.

Annunciator

Enables Cisco Unified CallManager to play pre-recorded announcements (.wav files) and tones to Cisco Unified IP Phones, gateways, and other configurable devices. The Annunicator is an SCCP device.

Anonymous Call Block

Allows the phone user to reject calls from anonymous callers.

Application Servers

Allows you to maintain associations between Cisco Unified CallManager and off-cluster, external applications, such as Cisco Unity and Cisco Unity Connection, and to synchronize Cisco Unified CallManager systems and other applications.

Attendant Console

Allows you to answer and direct calls using the Cisco Unified CallManager Attendant Console. Features include Busy Lamp Field (BLF) for stations, Speed Dialing, call queuing (with Music On Hold (MOH)), Call Park, multiple attendant support, broadcast hunting, support for shared lines, multiple calls on line, direct transfer and join, and transfer to voicemail.

See also Operator Attendant.

Auto Answer

Allows the phone user to connect incoming calls automatically after one or two rings, without pressing a button or picking up the handset.

The phone user might use Auto Answer if the phone user receives a high volume of calls.

As the system administrator, you can enable Auto Answer to work with either the speakerphone or headset.

Associated buttons: **Headset**, **Speaker**

AutoAttendant

Allows callers to locate people in the phone user's organization without talking to a receptionist.

Auto Dial

Allows the phone user to choose from a list of previously dialed numbers that could match the number that the phone user is currently dialing.

As the phone user begins to dial on hook (without a dial tone), Auto Dial displays any previously dialed numbers from the Placed Calls log that match the entered string of digits. The phone user can choose a number from the Auto Dial list to place the call or can continue to dial without using the Auto Dial feature.

Auto Registration

Allows Cisco Unified CallManager to assign directory numbers (DNs) automatically to new phones as they connect to the Cisco Unified Communications IP telephony network.

Authorization Codes

Allows authorized phone users who are using another user's phone to enter a code to temporarily override the class of service for the other user's phone and make an outside call in accordance with the user's own class of service restrictions.

Automated Alternate Routing (AAR)

Automates rerouting of calls through the Public Switched Telephone Network (PSTN) or other networks by using an alternate number when Cisco Unified CallManager blocks the call due to insufficient location bandwidth. The caller does not need to hang up and redial the called party.

Auto-Registered Phones Support Registration

Allows the phone user to automatically register a phone with the network.

Auto Route Selection (ARS)

Routes calls over the Public Switched Telephone Network (PSTN) based on the preferred (generally the least expensive) route available at the time that the call is placed. Least Cost Routing is a more sophisticated version of this feature.

See also Least Cost Routing.

Automatic Call Distributor

Distributes incoming calls evenly across call center agents and provides supervisory functions and statistical data for system management. Used primarily in contact centers.

Automatic Identified Outward Dialing

Allows you to specify extension numbers instead of the main PBX number for outbound calls. Used for billing purposes.

Background Image Setting

Allows the phone user to change the background image that is displayed on the phone screen.

Associated button: **Settings**

Backlight Setting

Allows the phone user to enable or disable the backlight on the wireless phone screen.

Associated menu path: **Menu > Phone Settings**

Barge

Allows the phone user to be added to non-private calls on a shared line.

Barge features include *Barge* and *cBarge*:

- Barge adds the phone user to a call but does not convert the call into a conference.
- cBarge adds the phone user to a call and converts it into a conference, allowing the phone user and other parties to access conference features.

Typically, only one of these barge features is available to the phone user.

Associated softkeys: **Barge**, **cBarge**

See also Conference Features and Shared Line.

Basic Rate Interface (BRI)

Provides the basic ISDN service level with two B-channels for voice and data and one D-channel for signaling.

Bridging

Allows a specified number of phone users who share a line appearance to go off hook simultaneously and actively bridge on the line (talk at the same time).

Brightness Setting

Allows the phone user to control the brightness level of the phone screen.

Associated button: **Settings**

Broadcast Paging

Allows the phone user to broadcast announcements through overhead speakers.

Browse To Onboard Device Statistics

Allows the phone user to view statistics concerning quality of service (QoS) or phone performance directly from the phone using the **Settings** button.

Associated button: **Settings**

Bulk Administration

Allows you to assigns lines and phone features to phones and gateways in bulk.

Busy Lamp Field (BLF)

Allows the phone user to monitor the line state (in-use or idle) of a phone line associated with a speed-dial button, call log, or directory listing on the phone.

The busy lamp field (BLF) feature does not prevent dialing; the phone user can place a call to the line regardless of BLF status.

Call Admission Control

Ensures that voice quality of service (QoS) is maintained across constricted WAN links and automatically diverts calls to alternate public switched telephone network (PSTN) routes when WAN bandwidth is not available.

Call Back

Allows the phone user to initiate an audio and visual alert on the phone when a busy or unavailable party becomes available.

Associated softkey: **CallBack**

Call Blocking

Allows you to prevent unauthorized used of phone based on time of day or designated ranges of phone numbers, such as toll-based 1-900 numbers.

Call Coverage

Allows one phone user to answer, or cover, another user's unanswered call. Includes the following capabilities:

- Forwarding—Distinguishes configuration based on whether the call originator is an internal user or an external user.
- Call pickup—Redirects a call that is ringing on another phone to the phone user's own phone, so that the phone user can answer the call.

See also Call Pickup.

Call Detail Record (CDR) Analysis and Reporting

Allows you to generate web-based Call Analysis and Reporting (CAR) reports that are based on the call detail records (CDRs) and call management records (CMRs) that Cisco Unified CallManager collects. Processes the CDR and CMR flat files that the CDR repository service places in the CDR repository and uses the information to generate reports on voice quality, traffic, and billing.

See also Call Details Record and Call Management Record (CMR).

Call Details Record

Allows the phone user to view the called number, the number that placed the call, the date and time that the call was started, the time that it connected, and the time that it ended. Used with the Cisco Unified CallManager CDR Analysis and Reporting or other supported reporting tool.

Associated softkey: **Details**

See also Call Detail Record (CDR) Analysis and Reporting.

Call Display Restrictions

Allows you to choose the information that will be displayed for calling and/or connected lines, depending on the parties who are involved in the call. By using specific configuration settings in Cisco Unified CallManager, you can choose to present or restrict the display information for each call.

Caller ID

Allows the phone user to see caller identification, such as a phone number, name, or other descriptive text, on the phone screen.

Caller ID Blocking

Allows the phone user to block the phone number or e-mail address from phones that have caller identification enabled.

Call Forking

Allows the same incoming call to be received on many endpoints in parallel or serial without requiring intelligent line preservation or hunt group logic. Once the call is answered, a second device cannot interact with the call.

Call Forward All

Allows the phone user to redirect incoming calls to another number. The phone user can set up call forwarding directly on the phone (for the primary line only) or from the User Options web pages (for any line on the phone).

Associated softkeys: **CFwdALL**, **CFwdAll**

Associated button sequence: ****1**

Call Forwarding

Allows you to specify forwarding treatment of incoming calls:

- Configurable display—Configure the information that is displayed when a phone is forwarded. The information is configurable on a per line appearance basis.
- Call Forward Fixed—Permit call forwarding to a pre-assigned location.
- Call Forward No Answer—Forward calls to another directory number (DN) when the called extension does not answer after a predefined number of rings.
- Call Forward Off-Premise—Forwards calls to off-premise stations.

Call History

See Call Logs.

Calling Search Spaces

Determines the partitions that calling devices, including IP phones, softphones, and gateways, can search when attempting to complete a call.

See also Partitions.

Call Logs

Allows the phone user to view records of missed, received, and placed calls on the phone.

Associated buttons: **Directories, Menu**

Associated menu path (wireless phones only): **Menu > Call History**

See also Extension Mobility.

Call Management Record (CMR)

Allows you to view diagnostic information on measures such as jitter, lost packets, amount of data sent and received during the call, and latency through the Cisco Unified CallManager CDR Analysis and Reporting or another supported reporting tool.

See also Call Detail Record (CDR) Analysis and Reporting and Call Details Record.

Call Overview

Allows the phone user to change the display on the phone screen to show one call per line. The displayed call per line is either the active call, or if all calls are on hold, the held call with the longest duration.

Associated button: **Line**

Call Park

Allows the phone user to park (temporarily store) a call and then retrieve the call by using another phone in the Cisco Unified CallManager system.

Call Park can be useful if the phone user wants to transfer a call from the phone to a phone in a lab or conference room, for example.

Associated softkey: **Park**

Call Pickup

Allows the phone user to redirect a call that is ringing on another phone to the phone user's own phone, so that the phone user can answer the call.

Call pickup can be useful for sharing call-handling tasks with coworkers.

Call pickup features include *Pickup*, *GPickup, and OPickup*:

Pickup allows the phone user to answer a call that is ringing on another phone within the phone user's "group" (a collection of extensions that you define).

- GPickup allows the phone user to answer a call ringing on a phone in another group.
- OPickup allows the phone user to answer a call ringing on a phone in another group that is associated with the phone user's group.

Associated softkeys: **PickUp**, **GPickUp, OPickUp**

Call Reporting Tool

Allows you to view information on quality of service, traffic, user call volume, billing, and gateways. Same as the Call Detail Record (CDR) and the Analysis and Reporting (CAR) tool.

See also Call Details Record and Call Management Record (CMR).

Call Screening

Allows you to prevent specified incoming calls from ringing the endpoint.

Call Status Per Line

Allows users with shared line appearances to see the call status of each shared line using an icon, an illuminated line button, or both.

Call Transfer Restrictions

Allow you to block call transfers between external parties by setting service parameters and configuring gateways, trunks, and route patterns.

Call Waiting

Alerts the phone user to incoming call while the phone user is on another call.

Call waiting provides an audio alert and displays incoming call information (visual alert) on the phone screen.

Associated softkey: **Answer**

See also New Call Indicator Setting.

cBarge

See Barge.

Cisco Unified Call Manager Assistant

Enables managers and their assistants to work together effectively through call routing service, enhancements to phone capabilities for the manager and assistant, and assistant console interfaces. Cisco Unified CallManager Assistant supports proxy line and shared line simultaneously in a cluster.

Class of Service

Quantifies the priority of the packets sent from an endpoint.

Client Matter Code

Allows the phone user to enter a billing or tracking code when placing a call.

See also Forced Authorization Code.

Comfort Noise Generation

Provides a low level of noise on the call to reassure callers that they are still connected during the periods when neither caller is talking.

Conference Features

Allows the phone user to talk simultaneously with multiple parties.

Conference features include *Ad Hoc Conference*, *cBarge*, *Join*, and *Meet-Me conference*:

- Ad hoc conference (or conference) allows the phone user to initiate a conference by calling each participant.
- cBarge allows the phone user to establish a conference by the phone user to a call on a shared phone line.
- Join allows the phone user to connect current callers who are on a single line by creating a conference call.
- Meet-Me allows the phone user to call a predetermined number at a scheduled time to host or join a conference.

Associated softkeys: **Confrn**, **Conference**, **Join**, **cBarge**, **MeetMe**

See also Ad Hoc Conference - Advanced, Join, Meet-Me Conference, Remove Conference Participants, and View Conference List.

Contrast Setting

Allows the phone user to adjust the contrast on the phone screen.

Associated button: **Settings**

Corporate Directory

Allows the phone user to use the phone users' own phone to search for coworkers' numbers.

The phone user can press a softkey or button to place a call to the phone number in the directory listing. If the phone user is on another call at the time, the phone might offer a menu of options for handling the first call before dialing the number in the directory listing.

Associated buttons: **Directories**, **Menu**

DHCP Support

Allows Cisco Unified IP Phones that are connected to the customer's data or voice Ethernet network to dynamically obtain their IP addresses and configuration information from a DHCP server.

Dependency Records

Allows you to find related Cisco Unified CallManager records and information.

Device Mobility

Allows you to track the home location and determine the identity and roaming properties of devices as they move from one location to another.

Device Pools

Allows you to define sets of common characteristics for phones and manage the phones that share the common characteristics as a single entity.

Device Profile

Allows you to specify the set of attributes (services and/or features) that are associated with a particular phone device.

Device Security Mode

Allows you to specify the level of security provided for the phone.

Dial Plan Partitioning

Allows you to implement call routing by dividing route plans into logical subsets based on organization, location, and call type. Also allows you to assign different classes of service to users.

See also Unified Dial Plan.

Dialed Number Analyzer

Allows you to display the call classification that is configured for the gateway and router pattern when performing digit analysis on a gateway.

Dialed Number Identification Service

Allows the called party to see the number that was dialed by the originator.

Dial Rules

Provides for automatically stripping numbers from or add numbers to telephone numbers that a user dials, such as adding the digit 9 in front of a 7-digit telephone number to provide access to an outside line.

Directory Lookup

Allows you to automatically transform caller identification numbers into numbers that can be looked up in the corporate directory, such as transforming a 10-digit number into a 5-digit extension.

Directory Service

Allows the phone user to enter a name using the keypad and have the station number displayed for easier call processing.

Direct Transfer

Allows the phone user to connect two calls to each other (without remaining on the line).

Associated softkey: **DirTrfr**

See also Transfer.

Directed Call Park

Allows the phone user to transfer and store an active call to a directed call park number. Allows the phone user to retrieve a parked call from any phone in the network by dialing the retrieval prefix and directed call park number.

Directed Call Park works with the BLF feature (if available) to indicate whether the line associated with the directed call park number is in use or idle.

Associated button: **Call Park BLF**

See also Busy Lamp Field (BLF) and Transfer.

Disable Ringer

See Do Not Disturb, New Call Indicator Setting, Wireless Phone Profiles, and Vibration Alert.

Disable Touchscreen

Allows the phone user to disable the touchscreen for cleaning (for phones with touch-sensitive phone screens only).

Associated button: **Display**

Disaster Recovery System

Provides full data backup and restore capabilities for all servers in a Cisco Unified CallManager cluster.

Distinctive Ring

See New Call Indicator Setting and Ring Tone Setting.

Do Not Disturb

Allows the phone user to block incoming calls from ringing on the phone.

When DND and Call Forward All are both enabled, Call Forward All takes precedence (incoming calls are forwarded).

Associated button: **Settings**

Associated softkey: **DND**

See also Do Not Disturb, Wireless Phone Profiles, and Vibration Alert.

Door Phone

Allows an intercom box or a combination intercom box/electronic door lock installed at an outside door to connect directly to a specific station or to the system.

Drop Conference Party

See Remove Conference Participants.

Dual Tone Multi Frequency

Refers to the standard touch tone sounds generated by the a telephone or PBX. Dual Tone Multi Frequency (DTMF) uses two tones to represent each key on the touch pad. When any key is pressed, both the tone of the column and the tone of the row are generated.

Edit Dial

Allows the phone user to edit a number that is displayed in a call record before dialing the number.

The phone user can use Edit Dial to add a prefix, for example, to a phone number in one of your call logs.

Associated softkey: **EditDial**.

See also Call Logs and Corporate Directory.

Enhanced 911 (E911)

Extends the basic 911 emergency call standard by providing additional information to emergency dispatchers regarding wireless 911 calls.

Extension Mobility

Allows the phone user to temporarily apply the phone user's phone number and user profile settings to a shared Cisco Unified IP Phone by logging into the Extension Mobility service on that phone.

Extension Mobility can be useful if the phone user works from a variety of locations within the phone user's company or if the phone user shares a workspace with coworkers.

Associated button: **Services**

Fast Dial

Allows the phone user to select a Fast Dial code to place a call.

The phone user can set up Fast Dials directly on the phone or from the User Options web pages. The phone user can assign Fast Dial codes to phone numbers and to Personal Address Book entries.

Fast Dials can be useful if the phone user's phone model does not provide speed dial-buttons or if the phone user wants to configure more speed-dial numbers than the number of speed-dial buttons on the phone.

Associated button: **Directories**

See also Abbreviated Dialing, Personal Address Book (PAB), and Speed Dialing.

Flexible Station Numbering

Allows the phone user to establish an individualized numbering scheme for directory numbers, feature access codes, and trunk access codes.

Forced Authorization Code

Allows the phone user to enter an authorization code to place calls to certain numbers that you specify.

See also Client Matter Code.

Gatekeeper

Supports the H.225 Registration, Admission, and Status Protocol (RAS) message set that is used for call admission control, bandwidth allocation, and dial pattern resolution (call routing).

Group Call Pickup

See Call Pickup.

Headset Support

Allows the phone user to use a phone headset.

Associated button: **Headset**

Help System (Phone)

Allows the phone user to get on-the-spot information about phone features, buttons, and softkeys.

Associated buttons: **?** or **i** button

Help System (Cisco Unified CallManager)

Allows you to obtain assistance on Cisco Unified CallManager tasks from the administrative user interface.

Hold

Allows the phone user to move a connected call from an active state to a held state.

The phone user's phone allows only one call to be active at a time; other calls will be put on hold.

Associated button: **Hold**

Associated softkey: **Hold**

See also Resume.

Hold Reversion

Limits the amount of time that a phone user can put a call hold before the call reverts back to the original phone and the user is alerted.

Hot Line

Makes a direct connection to another specified phone when the phone user goes off hook.

Also known as Private Line Automated Ringdown (PLAR).

Hunt Groups

Allows you to manage a set of destination extensions (hunt group) that are used for incoming calls. Several different algorithms are available to determine the order in which members of the hunt group receive calls:

- First Available Hunt Group Member—Finds the first destination that is available.
- Longest Idle Hunt Group Member—Finds the member that has been idle for the longest period of time.
- Circular Hunting—Records the last number that received a call and directs the call to the next member of the hunt group.
- Broadcast Hunting—Places the call on hold, adds the call to the call queue, and displays the call in the Broadcast Calls window on attendant PCs. Any attendant can answer the call from the Broadcast Calls window.

See also Station Hunting.

Hunt List

Controls the search for available directory numbers (DNs) for incoming calls by ordering a set of line groups to determine when the line groups are accessed.

See also Line Groups.

Hunt Pilot

Comprises a string of digits (an address) and a set of associated digit manipulations that route calls to a hunt list.

See also Hunt List and Route Filters.

Immediate Divert

Allows the phone user to transfer a ringing, connected, or held call directly to the phone user's voice-messaging system.

Associated softkey: **iDivert**

Join

Allows the phone user to join two or more calls that are on one line to create a conference call. The phone user remains on the call.

Associated softkey: **Join**

See also Conference Features.

Keypad Lock

Allows the phone user to lock and unlock the wireless phone keypad.

Associated button: **#**

Last Number Redial

Allows the phone user to redial the last number that was dialed from the phone.

Least Cost Routing

Provides capabilities that are similar to Auto Route Selection (ARS), including routing by time of day, day of week, and digit translation.

See also Auto Route Selection (ARS).

Lightweight Directory Access Protocol (LDAP)

Provides a Lightweight Directory Access Protocol (LDAP) Version 3 directory interface to selected vendor LDAP directories.

Line Groups

Allows you to designate the order in which directory numbers (DNs) are chosen.

Line Text Label

Allows the phone user to create a text label that appears on the phone screen for each phone line.

This feature can be useful if the phone user has multiple lines on the phone.

The phone user can access the line label setting from the User Options web pages.

Locale Setting

Allows the phone user to change the language (locale) that the User Options web pages and/or phone screen use to display text.

The phone user can access locale settings from the User Options web pages.

Location Configuration

Allows you to regulate audio quality and video availability by limiting the amount of bandwidth that is available for audio and video calls over links between locations.

See also Region Configuration.

Log Out of Hunt Group

Allows the phone user to log out of a hunt group and temporarily block hunt group calls. Logging out of hunt groups does not prevent non-hunt group calls from ringing the phone user's phone.

See also Hunt Groups.

Associated softkey: **HLog**

Log Partition Monitoring

Allows you to configure thresholds to monitor the disk usage of the log partition on a server (or all servers in the cluster) using Alert Central in Real Time Monitoring Tool.

See also Real Time Monitoring Tool.

Malicious Call ID

Allows the phone user to notify you about suspicious or harassing calls.

Associated softkey: **MCID**

Manager Assistance

Provides shared line support with multiple managers per assistant and redundant service:

- Manager features—Immediate divert or transfer, do not disturb, divert all calls, call intercept, call filtering on calling line identification (CLID), intercom, speed dials, barge, direct transfer, and join.
- Assistant features—Call handling on behalf of a manager, manager status and calls, speed dials, people search in the Corporate/Cisco Unified Call Manager directory, calls on the user's own line, immediate divert or transfer, intercom, barge, privacy, multiple calls per line, direct transfer, join, send dual tone multi frequency (DTMF) digits from console, and message waiting indicator (MWI) status of managers' phone.

Media Resource Group

Allows you to manage media resources, so that all Cisco Unified CallManagers within a cluster can share the resources.

Media Termination Point

Allows Cisco Unified CallManager to relay calls that are routed through SIP or H.323 endpoints or gateways.

Meet-Me Conference

Allows the phone user to host a Meet-Me conference in which other participants call a predetermined number at a scheduled time.

Associated softkey: **MeetMe**

See also Conference Features.

Message Waiting Indicator

Allows the phone user to see if there is an incoming call or new voice message by looking at the message waiting light (or “lamp”) on the handset.

By default, the message waiting lamp glows steadily to indicate a new voice message. The phone user can specify a different message waiting lamp policy by changing the setting in the User Options web pages.

Similarly, the phone user can change how the lamp on the handset behaves when the phone rings.

See New Call Indicator Setting.

Missed Call List

Maintains a list of missed phone calls in a special directory.

Mobility Management

Allows the phone user to answer incoming calls on the desktop phone or cellular phone, pick up in-progress calls on the desktop phone or cellular phone without losing the connection, and originate enterprise calls from the cellular phone.

Multilevel Administration Access

Allows you to define levels of access (read and write or read-only) for each configuration option.

Multilevel Precedence and Preemption (MLPP)

Allows for prioritization of calls within the phone user’s phone system. This feature is useful for making and receiving urgent or critical calls and will preempt a low-priority call for a higher priority call.

Multiparty Conference

Provides support for multiparty conferences that are established by the station or attendant.

Multiple Calls per Line Appearance

Allows each line on the phone user's phone to support multiple calls.

The typical default configuration specifies four calls per phone line, but you can adjust this setting.

Only one call can be active at any time; other calls will be on hold.

Multiple Lines per Phone

Allows the phone user to make and receive calls using more than one phone line on the same phone.

You assign one or more phone lines (directory numbers) to the phone user's phone.

Multiple Trunk Groups

Allows you to assign Public Switched Telephone Network (PSTN) trunk lines to pools or groups with similar functions.

Music On Hold (MOH)

Provides music supplied from a streaming source to a caller who has been placed on hold by another caller. The integrated Music On Hold feature comprises the media server, data base administration, call control, media resource manager, and media control functional areas.

Mute

Allows the phone user to disable the audio input for the phone user's handset, headset, speakerphone, and external microphone, so that the phone user can hear other parties on the call but cannot be heard by the other parties.

Associated button: **Mute**

Associated softkeys: **MuteOn**, **MuteOff**

Network Time Protocol (NTP) Reference

Allows you to configure the phone Network Time Protocol (NTP) references to ensure that the SIP phone obtains its date and time from the NTP server.

New Call Indicator Setting

Allows the phone user to specify the call indicator that the phone uses to indicate an incoming call on each line, depending on whether the phone is in use or idle.

Ring indicator settings include ring once, flash only, beep only, and do nothing (no ring).

The call indicator that the phone uses when it is in use is also known as the call-waiting indicator.

Changes to the call indicator setting do not affect the caller ID and status information that appears on the phone screen for an incoming call.

The phone user can access the call indicator setting from the User Options web pages. You must make this feature available to the phone user.

See also Call Waiting, Ring Tone Setting.

Off-Premise Extension

Provides off-site stations with the same capabilities and features as on-site stations.

On-hook Dialing

See Pre-Dial.

On-hook Transfer

Allows the phone user to hang up to complete the transfer of a call.

See also Transfer.

Online Help

See Help System (Phone).

Operator Attendant

Provides operator functionality through the Attendant Console.

Outbound Call Blocking

Allows you to restrict calls to certain predefined ranges of numbers (for example, by area code).

Outgoing Call Restrictions

Allows you to prevent specified stations from making outgoing calls, or restricts station calls to specific locations. You can implement these restrictions by placing stations in a common class of service.

Partitions

Allows you to specify logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics.

See also Calling Search Spaces.

Password Setting

Allows the phone user to change the password from the User Options web pages.

Performance Counters

Allows you to monitor system activities and usage.

See also Performance Objects and Real Time Monitoring Tool.

Performance Objects

Provides a set of counters that report on specific processes or applications.

See also Performance Counters and Real Time Monitoring Tool.

Personal Address Book (PAB)

Allows the phone user to create a directory of personal contacts to access directly from the phone or from the User Options web pages.

Associated button: **Directories**

See also Personal Directory.

Personal Address Book Synchronizer

Allows the phone user to synchronize data that is stored in the Microsoft Windows, Microsoft Outlook, or Microsoft Outlook Express address book(s) with the Cisco Unified CallManager directory and the Personal Address Book.

See also Personal Address Book (PAB).

Personal Directory

Allows the phone user to set up and use Personal Directory features (Personal Address Book and Fast Dials) directly on the phone or from the User Options web pages.

Associated button: **Directories**

See also Fast Dial and Personal Address Book (PAB).

Personalized Ring Tones

Allows the phone user to change ring tones to personalize the phone.

Phone Book

Allows the phone user to create and use a personal phone book on the wireless phone.

Associated softkey: **PhBook**

See also Personal Address Book (PAB).

Phone Button Templates

Allow you to assign a common phone button configuration to a large number of phones.

Phone Help

See Help System (Phone).

Phone Lock

Allows the phone user to lock the wireless phone.

Associated menu path: **Menu > Phone Settings**

Phone Services

Allows the phone user to subscribe to phone services from the User Options web pages.

After subscribing to a service, the phone user can access it from the phone using the services menu or a service URL button. After the phone user makes the phone services available, the User Options web pages can be used to assign services to the programmable buttons.

Phone services can include features, network data, and web-based information (such as stock quotes and movie listings).

Associated buttons: **Services**, **Svcs**

See also Service URL Button.

Phone Screen

Allows the phone user's phone to display features, call activity, caller ID, and other information.

PIN Setting

Allows the phone user to change the PIN setting from the User Options web pages.

Placed Call List

Allows you to maintain a list of placed phone calls in a special directory within the phone.

Power Failure Transfer

Allows a small number of central office trunk lines to be directly connected to certain telephones in the event of a system or commercial power failure.

Pre-Dial

Allows the phone user to enter a phone number before getting a dial tone. The phone user goes off hook to complete the call.

Presence

Allows the phone user to monitor the real-time status of another user at a directory number or SIP identifier.

Privacy

Allows the phone user to prevent coworkers who share the phone user's line from adding themselves to the phone user's calls and from viewing information on their phone screens about the phone user's calls.

Associated button: **Privacy**

See also Shared Line.

Private Line

Allows you to assign a Public Switched Telephone Network (PSTN) trunk line to a specific button that appears only on certain telephones or provides access only to specified phone users.

Private Line Automated Ringdown

Makes a direct connection to another predefined phone when the phone user goes off hook. Also known as Hot Line.

Programmable Buttons

Allows the phone user to access the following functions:

- Phone lines (line buttons)
- Speed-dial numbers (speed-dial buttons)
- Web-based phone services (for example, a corporate calendar button)
- Phone features (for example, a Privacy button)

You can configure programmable buttons for the phone user's phone and determine how many lines (and therefore, line buttons) are available on the phone.

Using the User Options web pages, the phone user can assign speed-dial numbers or phone services to available programmable buttons.

See also Speed Dialing and Service URL Button.

Public Service Telephone Network (PSTN) Failover

Provides failover to the PSTN if the WAN is busy or overloaded.

Quality of Service Statistics

Allows the phone user to view quality of service (QoS) statistics on a per call basis by means of HTTP browsing on the phone.

Quality Reporting Tool

Allows the phone user to submit call quality information to you.

Associated softkey: **QRT**

Q.Signaling

Provides interoperability and feature transparency among different types of telecommunications equipment. Based on the QSIG standard.

Real Time Monitoring Tool

Allows you to monitor device status, system performance, device discovery, and CTI applications in the Cisco Unified CallManager cluster using a client side plug-in application.

Recent Calls List

Allows the phone user to access missed, placed, and received calls using the directory function on the phone.

Recorded Messages

Allows callers to be connected to recorded announcements in conjunction with night service, station hunting, automatic call distribution, and auto wakeup.

Redial

Allows the phone user to call the most recently dialed phone number by pressing a button.

Associated softkey: **Redial**

Region Configuration

Allows you to specify the bandwidth that is used for audio and video calls within a region and between existing regions.

See also Location Configuration.

Remove Conference Participants

Allows the phone conference initiator to drop participants from the conference call by using *Remove* or *Remove Last Conference Participant*:

- *Remove* drops the selected participant.
- *Remove Last Conference Participant* drops the most recently added participant.

Associated softkeys: **Remove**, **RmLstC**

Repeat Last Number Dialed

Allows the phone user to redial the last number dialed by pressing a single key (or by dialing an access code).

Restore Phone Settings

Allows the phone user to restore phone settings (contrast, ring type, and volume) to previous values.

Associated softkeys: **Restore**, **Default**

Resume

Allows the phone user to resume a call that the phone user had put on hold.

Associated button: **Hold**

Associated softkey: **Resume**

See also Hold.

Ringer Settings

See New Call Indicator Setting and Ring Tone Setting.

Ringing Line Preference

Allows automatic selection of the ringing line when the phone user picks up the handset.

Ring Tone Setting

Allows the phone user to change the ring sound for each phone line.

Associated buttons: **Settings**, **Menu**

Associated menu path (wireless phones only): **Menu > Phone Settings**

See also New Call Indicator Setting.

Roles

Allows Cisco Unified CallManager administrators who have full administration privilege (access) to configure end users and application users with different levels of privilege.

Route Filters

Allows you to determine which route patterns or hunt pilots your users can dial; for example, whether phone users can manually choose a long-distance carrier (by dialing 101 plus a carrier access code).

See also Hunt Pilot.

Route Groups

Allows you to designate the order in which gateways and trunks are selected for outgoing calls.

See also Route Lists.

Route Lists

Allows you to associate a set of route groups with a specified priority order.

See also Route Groups.

Route Patterns

Allows you to specify a string of digits (an address) and a set of associated digit manipulations that route calls to a route list or a gateway. Used with Route Filters and Route Lists.

Route Plan Reports

Provides you with information about all assigned and unassigned directory numbers (DN), call park numbers, call pickup numbers, conference numbers, route patterns, translation patterns, message-waiting indicators, voice mail ports, and Cisco Unified CallManager Attendant Console pilot numbers in the system.

Route Point

Allows you designate a virtual device that can receive multiple, simultaneous calls for controlled redirection by Cisco Unified CallManager.

Resource Reservation Protocol (RSVP)

Provides call admission control through a resource-reservation, transport-level protocol designed to reserve resources in IP networks. Used as an alternative to location-based call admission control (CAC)

See also Call Admission Control.

Saved Number Redial

Allows the phone user to manually save the last number dialed so that it can be redialed at a later time by pressing a key (or dialing an access code).

Security

Provides the following options for Cisco Unified IP Phone network and management security:

- Configurable operation modes: non-secure or secure
- Device authentication: Embedded X.509v3 certificate in new model phones.
- Data Integrity: Support for Transport Layer Security (TLS) with the NULL-SHA cipher.
- Privacy: Encryption of signaling and media.
- Phone Security: TFTP files (configuration and firmware loads) signed with the self-signed certificate of the TFTP server. Support for disabling HTTP and Telnet on phone.
- Encryption: Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7960, MGCP gateways, SRST,HTTPS for secure administration of Cisco Unified CallManager, Locally significant certificates on Cisco Unified IP Phone 7970G systems, SSL for secure transport of user information between Cisco Unified CallManager applications and directories.

Serviceability Report Archive.

Allows you to view daily reports on device statistics, server statistics, service statistics, call activities, and alert summaries.

Service URL Button

Allows the phone user to access a service from a programmable button on the phone, rather than by using the Services button and Services menu.

After you make the service URL buttons available, the phone user can go to the User Options web pages and assign services to the programmable buttons.

See also Phone Services and Programmable Buttons.

Session Initiation Protocol (SIP)

Provides call setup, routing, authentication, and multimedia support for IP Phones using the standard Session Initiation Protocol (SIP).

Shared Line

Allows the phone user to use multiple phones that share the same phone number. Allows the phone user to share a phone number with a coworker.

Shared lines can use special features such as Barge, Privacy, and Cisco Unified CallManager Assistant.

See also Barge, Privacy, and Cisco Unified Call Manager Assistant.

Silence Suppression

Preserves WAN bandwidth by ceasing to transmit of voice packets when callers are not speaking (periods of silence).

Simple Message Desk Interface (SMDI)

Provides industry standard protocol support for connecting PBXs to voice mail systems. Simple Message Desk Interface (SDMI) passes information, such as the called party extension and reason code, to the voice mail system.

Simple Network Management Protocol (SNMP)

Allows you to manage network performance, find and solve network problems, and plan for network growth through the exchange of management information between network devices. The supported Simple Network Management Protocol (SNMP) versions depend upon the version of Cisco Unified CallManager that is running.

Softkeys

Allows the phone user to select phone features through context-sensitive displays and keys on selected phones.

Speaker Mode (listen-only)

Allows the phone user to listen without using the handset (hand-free).

Associated softkey: **Monitor**

See also Speakerphone Mode.

Speakerphone Mode

Allows the phone user to talk and listen hands-free (without using a handset or headset).

Associated button: **Speaker**

Speed Dialing

Allows the phone user to enter an index code, press a button, or select a phone screen item to place a call, rather than dialing the number manually.

The phone user can use the User Options web pages to assign a speed-dial number to a programmable phone button or to an Abbreviated Dialing index code (1-99).

Associated buttons: speed-dial button, assigned keypad button

See also Abbreviated Dialing and Fast Dial.

Station Hunting

Allows incoming calls to search through a group of assigned stations until an available station is located.

See also Hunt Groups.

Station Message Detail Recording

Allows you to generate reports about time, duration, and the numbers dialed for incoming and outgoing calls.

Station Volume Controls

Allows the phone user to adjust the volume for incoming call ringer and phone speaker. It also allows the phone user to adjust the outgoing volume of the phone microphone on a hands-free call.

Survivable Remote Site Telephony (SRST)

Provides telephony service to IP phones at a remote site in the event of a WAN outage and loss of connection to a centralized Cisco Unified CallManager server. Permits calls between IP phones at the remote site, allows calls from the PSTN to reach the IP phones, and allows calls from the IP phones to reach the external world through the PSTN.

Syslog Support

Provides a means for caching debug information in a file so that a diagnostic tool can retrieve the information from the phone's memory. This feature is for phones that do not support Simple Network Management Protocol (SNMP).

See also Simple Network Management Protocol (SNMP).

T.38 fax support (H.323 only)

Provides support for the T.38 for fax codec when using H.323 gateways.

T1/E1 Digital Trunk Interface

Provides PBX support for direct digital trunk interfaces.

Telnet Relay Application

Provides Cisco Service Engineers (CSEs) with transparent firewall access to Cisco Unified CallManager servers at the customer's site.

Tenant Service

Allows more than one organization, or tenant, to share the same PBX or Cisco Unified CallManager system. Each tenant is restricted to its own Public Switched Telephone Network (PSTN) trunks, attendant consoles, extension numbers, and incoming calls.

Time of Day, Day of Week, Day of Year Routing / Restrictions

Allows you to assign time schedules to partitions to determine when a phone, gateway, translation pattern, or route pattern is reached.

Time Zones

Allows you to configure different time zones for sets of devices.

Toll Fraud

Initiates ad hoc drops of conferences when the conference originator hangs up or all internal callers hang up. Also blocks transfers from external trunks or gateways to external trunks or gateways.

See also Call Transfer Restrictions.

Toll Restrictions

Allows you to prevent specified phones from making long distance (toll) calls.

Touchscreen

Allows the phone user to press the phone screen to choose menu items, softkeys, and feature tabs (on Cisco Unified IP Phones with touch-sensitive phone screens).

Trace

Assists in troubleshooting Cisco Unified CallManager problems. You can use the trace and log central options in the Real-Time Monitoring Tool to view the information.

See also Real Time Monitoring Tool.

Traffic Measurement

Allows you to generate statistical usage reports about the number of calls and duration of calls placed on different trunks or trunk groups, attendant consoles, and all-trunks-busy conditions. You can also track the frequency of feature use.

Transcoder Resources

Provides translation of real time protocol (RTP) streams from one codec format into another.

Transfer

Allows the phone user to redirect a connected call from the phone user's phone to another number. Transfer features include *Direct Transfer* and *Transfer*:

- Direct Transfer allows the phone user to transfer two calls to each other (without remaining on the line).
- Transfer allows the phone user to redirect a single call to a new number (with or without consulting the transfer recipient).

Associated softkeys: **DirTrfr**, **Transf**, **Transfer**, **Trnsfer**,

See also Direct Transfer and On-hook Transfer.

Trunk Groups

Allows you to define logical grouping of trunks and categorize them by purpose, such as local trunks, long distance trunks, and direct inward dialing (DID) trunks.

Trunk-to-Trunk Connections

Allows the phone user to establish a conference connection between two outside lines and then drop out of the conversation while the conference continues.

Unified Dial Plan

Allows you to set up a flexible plan for extension-to-extension dialing between systems. For example, you could set up a four-digit unified dial plan to connect users at a location with 3XXX extensions to those at another location with 4XXX extensions.

See also Dial Plan Partitioning.

User Options Web Pages

Allows the phone user to use the phone user's computer to control features, settings, and services for the phone. For example, the phone user can set up speed-dial buttons from the User Options web pages.

You can provide a User Options URL and login information.

Vibration Alert

Allows the phone user to turn the vibration alert on or off on the wireless phone.

Associated button: *

Video Display Mode

Allows the phone user to select the video display mode for viewing a video conference.

Supported modes depend on how you configure the phone system.

Associated softkey: **VidMode**

Video Support

Allows the phone user to make video calls, assuming that compatible equipment (such as a video phone or camera, and video software), is available.

Video support is indicated by an icon on the phone.

View Conference List

Allows the phone user to view current participants in a conference call.

Associated softkeys: **ConfList**, **ConfLis**

See also Conference Features, Remove Conference Participants.

Viewing Angle Settings

Allows the phone user to adjust the phone screen to accommodate the phone user's viewing angle.

Associated button: **Settings**

Voice Activity Detection

Detects when a caller has started speaking after a period of silence and resumes sending packets. Used in conjunction with Silence Suppression.

Voice Messaging

Allows the phone user to access a voice-messaging service from the phone user's phone, if the voice-messaging service is available.

Associated buttons: **Messages**, **Menu**

Associated softkey: **Messages**

Associated menu path (wireless phones only): **Menu > Messages**

See also Message Waiting Indicator.

Voice Synthesizer

Allows you to set up synthesized electronic voice prompts or messages for use in conjunction with system programming or message waiting.

Volume Settings

Allows the phone user to adjust the volume level for the currently active audio device (handset, headset, or speaker). When no audio devices are active, pressing the Volume button adjusts the ringer volume.

Associated buttons: **Volume**, Up/Down arrows

Associated softkey: **Volume**

Video Telephony (VT) Advantage

Provides video telephony functionality to Cisco Unified IP Phones. VT Advantage comprises Cisco Unified Video Advantage software and Cisco VT Camera II, a video telephone USB camera.

WebDialer

Allows the phone user to make calls to directory contacts by clicking items in a web browser.

Associated softkey: **Dial**

Wideband Audio Codec Support

Provides voice quality that approaches high fidelity reproduction. Wideband audio code support is based on the G.722 standard.

Wireless Phone Profiles

Allows the phone user to change settings and user profiles on a wireless phone.

User profiles allow the phone user to change speaker and ringer volume settings, keypad tones, and low battery indicators.

Network profiles allow the phone user to select from a list of profiles with settings for different wireless LAN sites.

Associated menu path: **Menu > Profiles**

XML Support

Allows you to display eXtensible Markup Language XML-formatted information on the IP phone screen.