Cisco Unified CME Features Roadmap

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This roadmap lists the features documented in the *Cisco Unified Communications Manager Express System Administrator Guide* and maps them to the modules in which they appear.

Feature and Release Support

Table 1-1 lists the Cisco Unified Communications Manager Express (Cisco Unified CME) version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified CME software also support that feature. Only features that were introduced or modified in Cisco Unified CME 4.0 or a later version appear in the table. *Not all features may be supported in your Cisco Unified CME software version*.

To determine the correct Cisco IOS release to support a specific Cisco Unified CME version, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/requirements/guide/33matrix.htm.

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1-1 Supported Cisco Unified CME Features

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 10.0			
10.0	Fast-Track Configuration Approach for Cisco Unified SIP IP Phones	Fast-Track Configuration feature provides a new configuration utility using which you can input the phone characteristics of a new SIP phone model.	Fast-Track Configuration Approach for Cisco Unified SIP IP Phones
	Cisco Jabber for Microsoft Windows	Cisco Jabber for Windows client is supported from Cisco Unified CME Release 10 onwards.	Cisco Jabber for Windows
	Cisco Unified CME-SRST License	Cisco Unified CME-SRST permanent license has been introduced along with the a new license package called Collaboration Professional Suite.	Licenses
	Secure SIP Trunk Support on Cisco Unified CME	Supports supplementary services in secure SRTP and SRTP fallback modes on SIP trunk of the SCCP Cisco Unified CME	Secure SIP Trunk Support on Cisco Unified CME
Cisco Unified CME 9.5			

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
9.5	Afterhours Pattern Blocking Support for Regular Expressions	Support for afterhours pattern blocking is extended to regular expression patterns for dial plans on Cisco Unified SIP and Cisco Unified SCCP IP phones.	After-Hours Pattern-Blocking Support for Regular Expressions
	Call Park Recall Enhancement	The **recall force** keyword is added to the **call-park system** command in telephony-service configuration mode to allow a user to force the recall or transfer of a parked call to the phone that put the call in park.	Call Park Recall Enhancement
	Display Support for Name of Called Voice Hunt Groups	The display of the name of the called voice-hunt-group pilot is supported by configuring the following command in voice hunt-group or ephone-hunt configuration mode:	Displaying Support for the Name of a Called Voice Hunt-Group
	Enhancement of Support for Hunt Group Agent Statistics	Support for hunt group agent statistics of Cisco Unified SCCP IP phones is enhanced to include the following information: • Total logged in time—On an hourly basis, displays the duration (in sec) since a specific agent logged into a hunt group. • Total logged out time—On an hourly basis, displays the duration (in sec) since a specific agent logged out of a hunt group.	Enhancement of Support for Ephone-Hunt Group Agent Statistics
	HTTPS Support in Cisco Unified CME	With Hypertext Transfer Protocol Secure (HTTPS) support in Cisco Unified CME 9.5 and later versions, these services can be invoked using an HTTPS connection from the phones to Cisco Unified CME.	HTTPS Provisioning For Cisco Unified IP Phones
	Localization Enhancements in Cisco Unified CME	Canadian French is supported as a user-defined locale on Cisco Unified SIP IP phones and Cisco Unified SCCP IP phones when the correct locale package is installed.	Localization Enhancements in Cisco Unified CME
	Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups	Local calls are prevented from being forwarded to the final destination using the **no forward local-calls to-final** command in parallel or sequential voice hunt-group configuration mode.	Preventing Local Call Forwarding to the Final Agent in a Voice Hunt-Groups
	Support for Voice Hunt Group Descriptions	a description can be specified for a voice hunt group using the **description** command in voice hunt-group configuration mode.	Support for Voice Hunt Group Descriptions
	Trunk to Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones	Trunk to trunk transfer blocking for toll bypass fraud prevention is supported on Cisco Unified Session Initiation Protocol (SIP) IP phones also.	Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones
Cisco Unified CME 9.0			

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
9.1	KEM Support for Cisco Unified 8961, 9951, and 9971 SIP IP Phones	Increases line key and feature key appearances, speed dials, or programmable buttons on Cisco Unified SIP IP phones.	Configuring Phones to Make Basic Calls
9.0	Cisco ATA-187	Supports T.38 fax relay and fax pass-through on Cisco ATA-187.	Configuring Cisco ATA Support
	Cisco Unified SIP IP Phones	Adds SIP support for the following phone types: • Cisco Unified 6901 and 6911 IP Phones • Cisco Unified 6921, 6941, 6945, and 6961 IP Phones • Cisco Unified 8941 and 8945 IP Phones	Phones in Cisco Unified CME

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Localization Enhancements for Cisco Unified SIP IP Phones	Provides the following enhanced localization support for Cisco Unified SIP IP phones: • Localization support for Cisco Unified 6941 and 6945 SIP IP Phones. • Locale installer that supports a single procedure for all Cisco Unified SIP IP phones.	Localization Support for Cisco Unified SIP IP Phones
	MIB Support for Extension Mobility in Cisco Unified SCCP IP Phones	Adds new MIB objects to monitor Cisco Unified SCCP IP Extension Mobility (EM) phones.	MIB Support for Extension Mobility in Cisco Unified SCCP IP Phones
	Mixed Shared Lines	Allows Cisco Unified SIP and SCCP IP phones to share a common directory number.	Mixed Shared Lines
	Multiple Calls Per Line	Overcomes the limitation on the maximum number of calls per line.	Multiple Calls Per Line
	My Phone Apps for Cisco Unified SIP IP Phones	Adds support for My Phone Apps feature on Cisco Unified SIP IP phones.	My Phone Apps for Cisco Unified SIP IP Phones
	Olson Timezone	Eliminates the need to update time zone commands or phone loads to accommodate a new country with a new time zone or an existing country whose city or state wants to change their time zone, using the **olsontimezone** command in either telephony-service or voice register global configuration mode.	Olson Timezones
	Paging Group Support for Cisco Unified SIP IP Phones	Allows you to specify a paging-dn tag and dial the paging extension number to page the Cisco Unified SIP IP phone associated with the paging-dn tag or paging group using the **paging-dn** command in voice register pool or voice register template configuration mode.	Paging Group Support for Cisco Unified SIP IP Phones
	Programmable Line Keys for Cisco Unified SIP IP Phones	Adds support for soft keys as programmable line keys on Cisco Unified 6911, 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP Phones.	Programmable Line Keys (PLK)
	Single Number Reach for Cisco Unified SIP IP Phones	Supports the following SNR features for Cisco Unified SIP IP phones: • Enable and disable the EM feature. • Manual pull back of a call on a mobile phone. • Send a call to a mobile PSTN phone. • Send a call to a mobile phone regardless of whether the SNR phone is the originating or the terminating side.	Single Number Reach for Cisco Unified SIP IP Phones
	Unsolicited Notify for Shared Line and Presence Events for Cisco Unified SIP IP Phones	Allows the Unsolicited Notify mechanism to reduce network traffic during Cisco Unified SIP IP phone registration using the bulk registration method.	Unsolicited Notify for Shared Line and Presence Events for Cisco Unified SIP IP Phones

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Virtual SNR DN for Cisco Unified SCCP IP Phones	Allows a call to be made to a virtual SNR DN and allows the SNR feature to be launched even when the SNR DN is not associated with any phone.	Virtual SNR DN for Cisco Unified SCCP IP Phones
	Voice and Video Hardware Conferencing	Enables a Cisco Unified SIP IP phone to act as the creator of ad-hoc and meet-me conferences with audio and video conferencing streams flowing from participating IP phones through the Cisco Unified CME.	Configuring Voice and Video Hardware Conferencing
	Voice Hunt Group Enhancements	Allows all ephone and voice hunt group call statistics to be written to a file using the **hunt-group statistics write-all** command.	Hunt Groups
Cisco Unified CME 8.8			
8.8	CTI CSTA Protocol Suite Enhancement	Enables the dial-via-office functionality from computer-based CSTA client applications and adds support to CSTA services and events.	CTI CSTA in Cisco Unified CME
	HFS Download Support for IP Phone Firmware and Configuration Files	Provides download support for SIP and SCCP IP phone firmware, scripts, midlets, and configuration files using the HTTP File-Fetch Server (HFS) infrastructure.	HFS Download Support for IP Phone Firmware and Configuration Files
	HTTPS Provisioning for Cisco Unified IP Phones	Allows you to import an IP phone's trusted certificate to an IP phone's CTL file using the i**mport certificate** command.	HTTPS support for an External Server
	Localization Enhancement	Adds localization support for Cisco Unified 3905 SIP and Cisco Unified 6945, 8941, and 8945 SCCP IP Phones.	System-Defined Locales
	Programmable Line Keys Enhancement	Adds support for soft keys as programmable line keys on Cisco Unified 6945, 8941, and 8945 SCCP IP Phones.	Programmable Line Keys (PLK)
	Real-Time Transport Protocol Call Information Display Enhancement	Allows you to display information on active RTP calls using the **show ephone rtp connections** command. The output from this command provides an overview of all the connections in the system, narrowing the criteria for debugging pulse code modulation and Cisco Unified CME packets without a sniffer.	Real-Time Transport Protocol Call Information Display Enhancement
	SIP Intercom	Adds intercom support to Cisco Unified SIP phones connected to a Cisco Unified CME system.	SIP Intercom
	Support for Cisco Unified 3905 SIP IP Phones	Adds support for SIP phones connected to a Cisco Unified CME system.	Phones in Cisco Unified CME
	Support for Cisco Unified 6945, 8941, and 8945 SCCP IP Phones	Adds support for SCCP phones connected to a Cisco Unified CME system.	Phones in Cisco Unified CME
Cisco Unified CME 8.6			
8.6	Bulk Registration Support for SIP Phones	Adds support for SIP phone bulk registration.	Bulk Registration Support for SIP Phones

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Clear Directory Entries in Missed/Placed/Received Calls List	Adds ability to clear phone call logs.	Clear Directory Entries
	Support for iPhone and iPod Touch Softphone Client	Adds support for SIP client software for iPhone and iPod Touch.	Support for Cisco Jabber
	Enhancement for Call-Forward Unregistered	Adds support for the CFU feature on SIP IP phones using the **call-forward b2bua unregistered** command under voice register dn tag.	Call Forward Unregistered
	Extension Mobility Support for SIP phone	Adds SIP phone support to extension mobility.	Extension Mobility for SIP Phones Enhancement
	Increase in the Number of Translation Rules	Increases the number of translation rules from 15 to 100 rules per translation rule table.	Defining Translation Rules for Callback-Number
	Localization Support for SIP IP Phones	Adds localization support for SIP IP phones.	Localization Support for Cisco Unified SIP IP Phones Multiple Locales SCCP: How to Configure Localization Support Configuring Multiple Locales
	SSL VPN SUPPORT on CUCME with DTLS	Adds enhanced SSL VPN support. Cisco Unified SCCP IP phones such as 7945, 7965, and 7975 located outside of the corporate network are able to register to Cisco Unified CME through an SSL VPN connection.	SSL VPN Support on Cisco Unified CME with DTLS Configuring SSL VPN Client with DTLS on Cisco Unified CME
	Support for 7926G Wireless SCCP IP Phone	Adds support for 7926G Wireless SCCP IP Phone.	Phones in Cisco Unified CME
	Video Conferencing and Transcoding	Allows you to use on-board Digital Signal Processor resources (PVDM3) to facilitate adhoc or meetme video conference calls.	Configuring Video Conferences Configuring Transcoding Resources
	Video and Camera Support for Cisco Unified IP Phones 8961, 9951, and 9971	Adds video support for IP phones 8961, 9951, and 9971.	SIP Endpoint Video and Camera Support for Cisco Unified IP Phones 8961, 9951, and 9971

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 8.5			
8.5	Customized Button Layout	Allows you to customize the display order of various button types on a phone using the button layout feature. The button layout feature allows you to customize the display of the following button types: • Line buttons • Speed Dial buttons • BLF Speed Dial buttons • Feature Buttons • ServiceURL buttons	Configuring Button Layout on SCCP Phones Configuring Button Layout on SIP Phones
	Customized Phone User Interface Services	Allows to customize the availability of individual service items such as Extension Mobility, My Phone Apps, and Single Number Reach (SNR) on a phone's user interface by assigning an individual service item to a button using the Programmable Line Key (PLK) **url-button** command.	Customized Phone User Interface Services
	E.164 Enhancements	Allows to present a phone number in + E.164 telephone numbering format. E.164 is an International Telecommunication Union (ITU-T) recommendation that defines the international public telecommunication numbering plan used in the PSTN and other data networks.	E.164 Enhancements
	Enhancement to Voice Hunt Group Restriction	Allows you to ignore the timeout value for voice hunt group member and the **call forward no answer** timer when **call forward noan** command is configured in a voice hunt group.	Enhancement to Voice Hunt Group Restriction
	Feature Policy Softkey Control	Allows you to control soft keys on the Cisco Unified SIP IP Phones 8961, 9951, and 9971 using the feature policy template. The feature policy template allows you to enable and disable a list of feature soft keys on Cisco Unified SIP IP Phones 8961, 9951, and 9971.	Feature Policy Softkey Control
	Forced Authorization Code	Allows you to manage call access and call accounting through the Forced Authorization Code (FAC) feature. The FAC feature regulates the type of call a certain caller may place and forces the caller to enter a valid authorization code on the phone before the call is placed. FAC allows you to track callers dialing non-toll-free numbers, long distance numbers, and also for accounting and billing purposes.	Configuring Forced Authorization Code
	Immediate Divert for SIP Phones	Allows you to immediately divert a call to a voice messaging system. You can divert a call to a voice messaging system by pressing the iDivert soft key on Cisco Unified SIP IP phones, such as 7940, 7040G, 7960 G, 7945, 7965, 7975, 8961, 9951, and 9971, with voice messaging systems (Cisco Unity Express or Cisco Unity).	SIP: Configuring Immediate Divert (iDivert) Soft Key

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Media Flow Around Support for SIP-SIP Trunk Calls(done added in cmesystm chapter)	Eliminates the need to terminate RTP and re-originate on Cisco Unified CME through the media flow around feature, reducing media switching latency and increasing the call handling capacity for Cisco Unified CME SIP trunks.	SIP: Enabling Media Flow Mode on SIP Trunks
	Overlap Dialing Support for SIP and SCCP IP Phones	Enables overlap dialing on SCCP and SIP IP phones such as, 7942, 7945, 7962, 7965, 7970, 7971, and 7975.	Overlap Dialing Support for SIP and SCCP IP Phones
	Park Monitor	Allows you to park a call and monitor the status of the parked call until the parked call is retrieved or abandoned. When a Cisco Unified SIP IP Phone 8961, 9951, or 9971 parks a call using the **park** soft key, the park monitoring feature monitors the status of the parked call.	Park Monitor
	Phone User Interface for BLF-Speed-Dial	Allows extension mobility (EM) users to configure dn-based Busy Lamp Field (BLF)-speed-dial settings directly on the phone through the Services feature button. BLF-speed-dial settings are added or modified (changed or deleted) on the phone using a menu available with the Services button.	Enabling BLF-Speed-Dial Menu
	Programmable Line Keys (PLK)	Allows you to program feature buttons or URL services button on phone's line keys. You can configure line keys as line buttons, speed dials, BLF speed dials, feature buttons, and URL buttons.	Programmable Line Keys (PLK)
	SNR Enhancements	Adds enhanced Single Number Reach feature for Cisco Unified CME: • Hardware Conference • Call Park, Call Pickup, and Call Retrieval • Answer Too Soon Timer • SNR Phone Stops Ringing After Mobile Phone Answers	SCCP: Configuring Single Number Reach Enhancements
	SSL VPN Client Support on SCCP IP Phones	Enables Secure Sockets Layer (SSL) Virtual Private Network (VPN) on SCCP IP phones such as 7945, 7965, and 7975.	Configuring SSL VPN Client for SCCP IP Phones
	XML API for Cisco Unified CME	Adds support for eXtensible Markup Language (XML) Application Programming Interface (API).	XML API for Cisco Unified CME
Cisco Unified CME 8.1			
8.1	Toll Fraud Prevention	Enables Toll Fraud Prevention on Cisco Unified CME to secure the Cisco Unified CME system against potential toll fraud exploitation by unauthorized users.	Configuring Toll Fraud Prevention
	Enhancements to SIP Phone Configuration	Allows you to verify SIP phone registration process, remove global registration parameters, and display details on phones that attempted to register with Cisco Unified CME and failed.	Cisco Unified CME Commands: show presence global through subnet.
	Support for Cisco Unified 6901 and 6911 SCCP IP Phones	Adds support for new SCCP IP phones 6901 and 6911.	Ephone-Type Parameters for Supported Phone Types

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 8.0(1)			
8.0	Cancel Call Waiting	Enables an SCCP phone user to disable Call Waiting for a call they originate.	Configuring Call Coverage Features
	CTI CSTA Protocol Suite	Allows computer-based CSTA client applications, such as a Microsoft Office Communicator (MOC) client, to monitor and control the Cisco Unified CME system to enable programmatic control of SCCP telephony devices registered in Cisco Unified CME.	Configuring CTI CSTA Protocol Suite
	IPv6 Support for SCCP Endpoints	Adds IPv6 support for SCCP phones. SCCP Phones can interact with and support any SCCP devices that support IPv4 only or both IPv4 and IPv6 (dual-stack).	Configuring IP Phones in IPv4, IPv6, or Dual Stack Mode
	Logical Partitioning Class of Restriction (LPCOR)	Enables a single directory number on an IP or analog phone that is registered to Cisco Unified CME to connect to both PSTN and VoIP calls according to restrictions specified by Telecom Regulatory Authority of India (TRAI) regulations.	Call Restriction Regulations
	MLPP enhancements	Adds enhanced Multilevel Priority and Preemption (MLPP) features for Cisco Unified CME including: • Additional MLPP announcements for isolated code (ICA), unauthorized precedence level (UPA), loss of C2 features (LOC2), and vacant code (VCA) • Multiple service domains for the Defense Switched Network (DSN) and Defense Red Switched Network (DRSN) • Route codes and service digits in dialing formats • Support for supplementary services, such as Three-Way Conferencing, Call Pickup, and Cancel Call Waiting on Analog FXS ports	Configuring MLPP
	Music On Hold Enhancement	Adds support for Music on Hold from different media sources.	Configuring Music on Hold Groups to Support Different Media Sources
	Secure IP Phone (IP-STE) Support	Adds support for secure IP Phone, IP-STE.	Secure IP Phone (IP-STE) Support

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 7.1			
7.1	Autoconfiguration of Cisco VG202, VG204, and VG224	Allows you to automatically configure the Cisco VG202, VG204, and VG224 Analog Phone Gateway from Cisco Unified CME.	Configuring Phones to Make Basic Calls
	Barge and cBarge for SIP Phones	Enables phone users to join a call on a SIP shared-line directory number.	Configuring Barge and Privacy
	BLF Monitoring of Ephone-DNs with DnD, Call Park, Paging, and Conferencing	Provides Busy Lamp Field (BLF) indicators for directory numbers that become DND-enabled or are configured as call-park slots, paging numbers, or conference numbers.	Configuring Presence Service
	BLF Monitoring of Devices	Supports device-based BLF monitoring, allowing a watcher to monitor the status of a phone, not only a line on the phone.	Configuring Presence Service
	Busy Trigger and Channel Huntstop for SIP Phones	Provides a busy trigger and channel huntstop for directory numbers on SIP phones to prevent incoming calls from overloading the phone.	Configuring Phones to Make Basic Calls
	Call Park Enhancements	Adds Call Park features for SIP phones and enhances the Directed Call Park feature.	Configuring Call Park
	Call Pickup Enhancements	Adds Call Pickup features for SIP phones and enables users to perform Directed Call Pickup using the GPickUp soft key.	Configuring Call Coverage Features
	DND Enhancement for SIP phones	Modifies DND behavior so that the SIP phone flashes an alert to visually indicate an incoming call instead of ringing and the call can be answered if desired.	Configuring Do Not Disturb
	DSCP	Supports Differentiated Services Code Point (DSCP) packet marking for Cisco Unified IP phones.	Configuring System-Level Parameters
	Privacy for SIP phones	Enables phone users to block other users from seeing call information or barging into a call on a SIP shared-line directory number.	Configuring Barge and Privacy
	Shared-Line Directory Numbers	Adds shared-line directory numbers for SIP phones.	Configuring Phones to Make Basic Calls
	Single Number Reach (SNR)	Enables users to answer incoming calls on their desktop IP phone or at a remote destination, such as a mobile phone.	Configuring Single Number Reach (SNR)
	SIP Trunk Video Support for SCCP Endpoints	Supports video calls between SCCP endpoints across different Cisco Unified CME routers connected through a SIP trunk. Supports H.264 codec for video calls.	Configuring Video Support
	Whisper Intercom	Provides a one-way voice path from the caller to the called party, regardless of whether the called party is busy or idle. The called phone automatically answers in speakerphone mode.	Configuring Intercom Lines

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 7.0(1)			
7.0(1)	**Note** Cisco Unified CME 7.0 includes the same features as Cisco Unified CME 4.3, which is renumbered to align with Cisco Unified Communications versions.		
	Cisco Unified CME Usability Enhancement	Automatically creates TFTP bindings using the enhanced **load** command if cnf location is router flash memory or router slot 0 memory.	How to Configure System-Level Parameters SCCP: Upgrading or Downgrading Phone Firmware Between Versions
		• Introduces locale installer that supports a single procedure for all SCCP IP phones. • Automatically creates the required TFTP aliases for localization. • Provides backward compatibility with the configuration method in Cisco Unified CME 7.0 and earlier versions.	Configuring Localization Support
	Cisco Unified CME TAPI Enhancement	Introduces a Cisco IOS command that disassociates and reestablishes a TAPI session that is in frozen state or out of synchronization.	Resetting and Restarting Phones
	Cisco Unity Express AXL Enhancement	Automatically synchronizes Cisco Unified CME and Cisco Unity Express passwords.	Integrating Voice Mail
	Cisco Unified IP Phones	Adds SCCP support for the following phone type:Cisco Unified Communications Manager Express 7.0/4.3 Supported Firmware, Platforms, Memory, and Voice Products • Cisco Unified Wireless IP Phone 7925	*Cisco Unified Communications Manager Express 7.0/4.3 Supported Firmware, Platforms, Memory, and Voice Products*
	VRF Support on Cisco Unified CME	Adds support for conferencing, transcoding, a RSVP components in Cisco Unified CME through a VRF; also allows soft phones and TAPI clients in data VRF resources to communicate with phones in a VRF voice gateway.	Configuring VRF Support
Cisco Unified CME 7.0/4.3			
7.0/4.3	Autoprovisioning Directory Numbers in SRST Fallback Mode	Allows you to specify whether Cisco Unified CME in SRST Fallback mode creates octo-line or dual-line directory numbers for ephone-dns that are “learned” automatically from the ephone configuration.	Configuring SRST Fallback Mode
	Barge	Enables phone users to join a call on a shared octo-line directory number by pressing the Cbarge soft key and converting the call to an ad hoc conference.	Configuring Barge and Privacy
	Call Transfer Recall	Enables a transferred call to return to the phone that initiated the transfer if the destination does not answer.	Configuring Call Transfer and Forwarding

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Cisco 3200 Series Mobile Access Router	Support for Cisco Unified CME on the Cisco 3200 Series Mobile Access Router was added.	Cisco Unified CME Overview
	Cisco Unified IP Phones	Adds SCCP support for the following phone types: • Cisco Unified IP Phone 7915 Expansion Module • Cisco Unified IP Phone 7916 Expansion Module • Cisco Unified IP Conference Station 7937 • Nokia E61 Adds SIP support for the following phone types: • Cisco Unified IP Phone 7942G and 7945G • Cisco Unified IP Phone 7962G and 7965G • Cisco Unified IP Phone 7975G	*Cisco Unified Communications Manager Express 7.0/4.3 Supported Firmware, Platforms, Memory, and Voice Products*
	Consultative Transfer Enhancements	Modifies the digit-collection process for consultative call transfers. After a phone user presses the Transfer soft key for a consultative transfer, a new consultative call leg is created and the Transfer soft key is not displayed again until the dialed digits of the transfer-to number are matched to a transfer pattern and consultative call leg is in alerting state.	Configuring Call Transfer and Forwarding
	Directory Search Enhancement	Increases the number of entries supported in a search results list from 32 to 240 when using the directory search feature.	Configuring Directory Services
	Extension Mobility Enhancement	Adds support for the following: • Automatic Logout, including: – Configurable time-of-day timers for automatically logging out all EM users. – Configurable idle-duration timer for logging out a single user from an idle EM phone. • Automatic Clear Call History when a user logs out from EM.	Configuring Extension Mobility
	Phone-Type Configuration	Allows you to dynamically add a new phone type to your configuration without upgrading your Cisco IOS software.	Configuring Phones to Make Basic Calls
	Live Record	Enables IP phone users to record a phone conversation when Cisco Unity Express is the voice mail system.	Integrating Voice Mail
	Maximum Ephones	Sets the maximum number of SCCP phones that can register to Cisco Unified CME using the **max-ephones** command, without limiting the number that can be configured. This enhancement also expands the maximum number of phones that can be configured to 1000.	Configuring System-Level Parameters
	Octo-Line Directory Numbers	Adds octo-line directory numbers that support up to eight active calls, both incoming and outgoing, on a single phone button. Unlike a dual-line directory number, an octo-line directory number can split its channels among other phones that share the directory number.	Configuring Phones to Make Basic Calls

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Privacy	Enables phone users to block other users from seeing call information or barging into a call on a shared octo-line directory number.	Configuring Barge and Privacy
	Push-to-Talk	Adds support for one-way Push-to-Talk (PTT) in Cisco Unified CME without requiring an external server to support the functionality. PTT is supported in firmware version 1.0.4 and later versions on Cisco Unified wireless IP phones with a thumb button.	SCCP: Configuring One-Way Push-to-Talk on Cisco Unified Wireless IP Phones
	Speed Dial/Fast Dial Phone User Interface	Allows IP phone users to configure their own speed-dial and fast-dial settings directly from the phone. Extension Mobility users can add or modify speed-dial settings in their user profile after logging in.	Configuring Speed Dial
	Transfer to Voice Mail	Allows a phone user to transfer a call directly to a voice-mail extension by pressing the TrnsfVM soft key.	Integrating Voice Mail
	Voice Hunt-Group Enhancements	Supports the following Voice Hunt Group features: • Call Forwarding to a Parallel Voice Hunt-Group (Blast Hunt Group). • Call Transfer to a Voice Hunt-Group. • Member of Voice Hunt-Group can be a SCCP phone, FXS analog phone, DS0-group, PRI-group, SIP phone, or SIP trunk.	Configuring Call Coverage Features
Cisco Unified CME 4.2(1)			
4.2(1)	Call Blocking Enhancements	Adds support for selective call blocking on IP phones and PSTN trunk lines.	Configuring Call Blocking
	Extension Assigner Synchronization	Provides support for automatically synchronizing configuration changes to backup systems	Creating Phone Configurations Using Extension Assigner
	Extension Mobility Phone User support in Cisco Unified CME GUI	Allows a phone user to use a name and password from an EM profile to log into the Cisco Unified CME GUI for configuring personal speed dials on an EM phone. EM options in the GUI cannot be accessed from the System Administrator or Customer Administrator login screens.	Accessing the Cisco Unified CME GUI

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 4.2			
4.2	Enhanced 911 Services	• Enables routing to the PSAP closest to the caller by assigning ERLs to zones. • Allows you to customize E911 services by defining a default ELIN, designated number for callback, expiry time for Last Caller table, and syslog messages for emergency calls. • Expands the E911 location information to include name and address. • Uses templates to assign ERLs to a group of phones. • Adds permanent call detail records.	Configuring Enhanced 911 Services
	Extension Mobility	Provides the benefit of phone mobility for end users by enabling the user to log into any local Cisco Unified IP phone that is enabled for extension mobility.	Configuring Extension Mobility
	Interoperability with Cisco Unified Contact Center Express (Cisco UCCX)	Enables interoperability between Cisco Unified CME and Cisco Customer Response Solutions (CRS) 5.0 and later versions with Cisco Unified Contact Center Express (Unified CCX), including Cisco Unified IP IVR, enhanced call processing, device and call monitoring, and unattended call transfers to multiple call center agents and basic extension mobility.	Configuring Interoperability with Cisco Unified CCX
	Media Encryption (SRTP) on Cisco Unified Communications Manager Express	Provides the following secure voice call capabilities: • Secure call control signaling and media streams in Cisco Unified CME networks using Secure Real-Time Transport Protocol (SRTP) and H.323 protocols. • Secure supplementary services for Cisco Unified CME networks using H.323 trunks. • Secure Cisco VG224 Analog Phone Gateway endpoints.	Configuring Security
Cisco Unified CME 4.1			
4.1	Call Forward All Synchronization	When a user enables Call Forward All on a SIP phone using the CfwdAll soft key, the uniform resource identifier (URI) for the service is sent to Cisco Unified CME. When Call Forward All is configured in Cisco Unified CME, the configuration is sent to the SIP phone which updates the CfwdAll soft key to indicate that Call forward All is enabled.	Configuring Call Transfer and Forwarding

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Cisco Unified IP Phones	Adds SCCP support for the following phones: • Cisco Unified IP Phone 7921G • Cisco Unified IP Phone 7942G and 7945G • Cisco Unified IP Phone 7962G and 7965G • Cisco Unified IP Phone 7975G Adds SIP support for the following phones: • Cisco Unified IP Phone 3911 • Cisco Unified IP Phone 3951 • Cisco Unified IP Phone 7911G • Cisco Unified IP Phone 7941G and 7941G-GE • Cisco Unified IP Phone 7961G and 7961G-GE • Cisco Unified IP Phone 7970G and 7971G-GE No additional configuration is required for these phones. They are supported in the appropriate Cisco IOS commands.	*Cisco Unified CME 4.1 Supported Firmware, Platforms, Memory, and Voice Products*
	Directory Services	Supports local directory and local speed dial features for SIP phones.	Configuring Directory Services
	Disabling SIP Supplementary Services for Call Forward and Call Transfer	Allows you to prevent REFER messages for call transfers and redirect responses for call forwarding from being sent by Cisco Unified CME if a destination gateway does not support supplementary services. Supports disabling of supplementary services if all endpoints use SCCP or all endpoints use SIP.	Configuring Call Transfer and Forwarding
	Enhanced 911 Services for Cisco Unified CME in SRST Fallback Mode	Routes callers dialing 911 to the correct location.	Configuring Enhanced 911 Services
	KPML	Allows Key Press Markup Language (KPML) to report SIP phone users' input digit by digit to Cisco Unified CME, which performs pattern recognition by matching a destination pattern to a dial peer as it collects the dialed digits.	Configuring Phones to Make Basic Calls
	Multi-Party Conferencing Enhancements	Provides the following enhancements: • Enhanced ad-hoc conferences are hardware-based and allow more than three parties. • Meet-me conferences consist of at least three parties dialing a meet-me conference number.	Configuring Conferencing

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Network Time Protocol	Allows SIP phones registered to a Cisco Unified CME router to synchronize to a Network Time Protocol (NTP) server, known as the clock master.	Defining Network Parameters
	Out-of-Dialog REFER	Allows remote applications to establish calls by sending an out-of-dialog REFER (OOD-R) message to Cisco Unified CME without an initial INVITE. After the REFER message is sent, the remainder of the call setup is independent of the application and the media stream does not flow through the application.	Defining Network Parameters
	Presence with BLF Status	Allows presence to support BLF notification features for speed dial buttons and directory call lists for missed calls, placed calls, and received calls. SIP and SCCP phones that support BLF speed-dial and BLF call-list features can subscribe to status notification for internal and external directory numbers.	Configuring Presence Service
	Restarting Phones	Allows SIP phones to quickly reset using the **restart** command. Phones contact the TFTP server for updated configuration information and re-register without contacting the DHCP server.	Resetting and Restarting Phones
	Session Transport	Allows TCP to be used as the transport protocol for supported SIP phones connected to Cisco Unified CME. Previously, only UDP was supported.	Configuring Phones to Make Basic Calls
	SIP Dial Plans	Enables SIP phones to perform local digit collection and recognize dial patterns as user input is collected using dial plans. After a pattern is recognized, the SIP phone sends an INVITE message to Cisco Unified CME to initiate the call.	Configuring Phones to Make Basic Calls
	Soft Keys	Allows you to customize the display and order of soft keys that appear on individual SIP phones during the connected, hold, idle, and seized call states.	Customizing Soft Keys
	Translation Rules	Allows SIP phones in a Cisco Unified CME system to support translation rules with functionality similar to phones running SCCP. Translation rules can be applied to incoming calls for directory numbers on a SIP phone.	Configuring Dialing Plans

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
Cisco Unified CME 4.0(3)			
4.0(3)	AMWI	Allows Cisco Unified IP Phone 7911 and Cisco Unified IP Phone 7931G to be configured to receive AMWI (Audible Message Line Indicator) and visual MWI notification from an external voice-messaging system.	Integrating Voice Mail
	Cisco Unified IP Phones	Adds support for the following phones: • Cisco Unified IP Phone 7906G • Cisco Unified IP Phone 7931G	*Cisco Unified CME 4.0(3) Supported Firmware, Platforms, Memory, and Voice Products*
	DSS	Introduces the DSS (Direct Station Select) feature that allows the phone user to press a single speed-dial line button to transfer an incoming call when the call is in the connected state. This feature is supported on all phones on which monitor line buttons for speed dial or speed-dial line buttons are configured.	Configuring Speed Dial
	Extension Assigner	Allows installation technicians to assign extension numbers to phones without administrative access to Cisco Unified CME, typically during the installation of new phones or the replacement of broken phones.	Creating Phone Configurations Using Extension Assigner
	Fax Relay	Introduces a SCCP-enhanced feature that adds support for Cisco Fax Relay and Super Group 3 (SG3) to G3 fax relay. The feature allows the fax stream between two SG3 fax machines to negotiate down to G3 speeds (less than 14.4 kbps) allowing SG3 fax machines to interoperate over fax relay with G3 fax machines.	Configuring Fax Relay
Cisco Unified CME 4.0(1)			
4.0(1)	Call Forwarding	**Automatic call forwarding during night service**—Ephone-dns (extensions) can be designated to automatically forward their calls to a specified number during the time that night service is in effect. **Blocking call forwarding of local calls**—Forwarding of local (internal) calls from other Cisco Unified CME ephones can be blocked. External calls will continue to be forwarded as specified by the configuration for the ephone-dns. **Selective call forwarding**—Call forwarding for busy and no-answer ephone-dns can be applied selectively based on the number that a caller dials for a particular ephone-dn: the primary number, the secondary number, or either of those numbers expanded through the use of a dial-plan pattern.	Configuring Call Transfer and Forwarding

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Call Park	**Call park blocked per ephone**—Individual ephones can be blocked from parking calls at call-park slots. **Call park redirect**—You can specify that calls use the H.450 or SIP Refer method of call forwarding or transfer to park calls and to pick up calls from park. **Dedicated call-park slots**—A private call-park slot can be configured for each ephone. **Direct pickup of parked call on monitored park slot** —A call that is parked on a monitored call-park slot can be picked up by pressing the assigned monitor button.	Configuring Call Park
	Call Pickup	**Directed call pickup disable**—The **no service directed-pickup** command globally disables directed call pickup and changes the action of the PickUp soft key to invoke local group pickup rather than directed call pickup.	Configuring Call Coverage Features
	Call Transfer	**Call transfer blocking**—When call transfers to phones outside the Cisco Unified CME system have been globally enabled, you can block them for individual ephones. **Call transfer destination digits limited**—When call transfers to phones outside the Cisco Unified CME system have been globally enabled, you can limit the number of digits that can be dialed when transferring a call. **transfer-system command**—The command default has been changed from the **blind** keyword to the **full-consult** keyword, making H.450.2 consultative transfer the default method. **QSIG supplementary services support**—H.450 supplementary services features allow Cisco Unified CME phones to use QSIG to interwork with PBX phones. IP phones can use a PBX message center with proper MWI notifications.	Configuring Call Transfer and Forwarding
	Cisco Unified IP Phones	Adds support for the following phones: • Cisco Unified IP Phone 7911G • Cisco Unified IP Phone 7941G and 7941G-GE • Cisco Unified IP Phone 7961G and 7961G-GE No additional configuration is required for these phones. They are supported in the appropriate Cisco IOS commands.	*Cisco Unified CME 4.0 Supported Firmware, Platforms, Memory, And Voice Products*
	Conferencing	**Drop last party or keep parties connected**—New options specify whether the last party that joined a conference can be dropped from the conference and whether the remaining two parties should be allowed to continue their connection after the conference initiator has left the conference. **Improved conference display**—A Cisco Unified IP phone that is connected to a three-way conference displays “Conference.” No special configuration is required.	Configuring Conferencing

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Feature Access Codes	**Feature Access Code (FAC) support**—The same FACs that are used by analog phones can be enabled for IP phones. In addition, standard FACs can be customized and aliases can be created to simplify the dialing of a FAC and any additional digits that are required to activate the feature.	Configuring Feature Access Codes
	Headset Auto-Answer	**Headset auto-answer—**When the headset key on a phone is activated, lines on the phone that are specified for headset auto-answer will automatically connect to incoming calls after playing an alerting tone to notify the phone user of the incoming call. This feature is available on Cisco Unified IP Phones 7940G, 7960G, 7970G, and 7971G-GE.	Configuring Headset Auto-Answer
	Hunt Groups	**Agent status control—**Hunt group agents can put their phones in a not-ready state to temporarily suspend the receiving of hunt group calls by using the HLog soft key. A new FAC can toggle ready and not-ready state. **Automatic agent not-ready status**—The criterion for placing a hunt group agent into not-ready status (previously called automatic logout) was changed. If an agent does not answer the number of consecutive hunt-group calls that you specify in the **auto logout** command, the agent's ephone-dn is put into not-ready status (logged out) and will not receive further hunt group calls. **Call hold statistics**—New fields describing the length of time that calls spend in the hold state are in the statistical reports for Cisco Unified CME B-ACD applications. See the **show ephone-hunt statistics** command and the **hunt-group report url** command in *Cisco Unified CME B-ACD and Tcl Call-Handling Applications.* **Dynamic hunt group membership—**Agents can join or leave a hunt group using standard or custom FACs when wildcard slots are configured for hunt groups and the agents' ephone-dns are authorized to join hunt groups. **Change in hops command default—**The maximum number of hops allowed by a hunt group is automatically adjusted to reflect the dynamically changing number of members. **Enhanced display of ephone hunt-group information**—A text string can be added to provide information in configuration output and to display on IP phones when a hunt-group call is ringing or answered or when all hunt-group members are logged out. **Local call forwarding restriction in sequential ephone hunt groups**—In sequential ephone-hunt groups, local (internal) calls to the hunt group can be prevented from being forwarded beyond the first ephone-dn in the hunt group.	Configuring Call Coverage Features

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Hunt Groups	**Longest-idle hunt group improvement**—The **from-ring** command specifies that on-hook time stamps should be updated when a call rings an agent and when a call is answered by an agent. **Maximum number of agents**—The maximum number of agents per hunt group has increased from 10 to 20. No special configuration is required. **Maximum number of hunt groups**—The maximum number of hunt groups per Cisco Unified CME system has increased from 10 to 100. No special configuration is required. **No-answer timeout enhancements**—No-answer timeouts in ephone hunt groups can be set individually for each ephone-dn in the list. A maximum cumulative no-answer timeout can be also be set. **Restricting presentation of calls to idle or on-hook phones**—The presentation of hunt group calls can be restricted to hunt-group members on phones that are idle or on-hook. This enhancement considers all lines on the phone, both members of the hunt group and nonmembers, when restricting presentation of hunt group calls. **Return to a secondary destination in an ephone hunt group after call park**—Calls parked by hunt group agents can be returned to a different entry point in the hunt group. **Return to transferring party on no answer in an ephone hunt group**—A call that was transferred into a hunt group and was not answered can be returned to the party that transferred it to the hunt group instead of being sent to voice mail or another final destination.	Configuring Call Coverage Features
	Localization	**Multiple user locales and network locales**—Up to five user and network locales are supported. **User-defined user locales and network locales**—User-defined locales can be added for supported phones.	Configuring Localization Support
	Music on Hold	**Music on hold (MOH) for internal calls**—Internal callers (those making calls between extensions in the same Cisco Unified CME system) hear music when they are on hold or are being transferred. The **mulitcast moh** command must be used to enable the flow of packets to the subnet on which the phones are located. Internal extensions that are connected through an analog voice gateway or through a WAN (remote extensions) do not hear MOH on internal calls. The ability to disable multicast MOH per phone was introduced, using the **no multicast-moh** command in ephone or ephone-template configuration mode.	Configuring Music on Hold

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Overlaid Ephone-dns	**Overlaid ephone-dns**—The maximum number of overlaid ephone-dns per ephone button has increased from 10 to 25. No special configuration is required. **Overlaid ephone-dn call-waiting display**—The number of waiting calls that can be displayed for overlaid ephone-dns that have call waiting configured has been increased to six for the Cisco IP Phone 7940G, 7941G, 7941G-GE, 7960G, 7961G, 7961G-GE, 7970G, and 7971G-GE. The overlaid ephone-dns must be configured on the phone using the **button** command and the **c** keyword. **Overlaid ephone-dn call overflow to other buttons**—One or more buttons can be dedicated to serve as expansion or overflow buttons for another button on the same Cisco Unified IP phone that has overlaid ephone-dns. A call to an overlay button that is busy with an active call will roll over to the next available expansion button.	Configuring Call Coverage Features
	Phone Support	**Cisco IP Communicator** is a software-based application that appears on a user's computer monitor as a graphical, display-based IP phone with a color screen, a key pad, feature buttons, and soft keys. Cisco Unified CME supports Cisco IP Communicator 2.0 and later versions. **Remote teleworker phone**—Teleworkers can connect remote phones over a WAN and be directly supported by Cisco Unified CME.	Configuring Phones to Make Basic Calls
	Ring Tones	**Distinctive ringing**—An extension's ring patterns can be set to distinguish among internal, external, and feature calls.	Configuring Ring Tones
	Security	**Cisco Unified CME phone authentication** is a security infrastructure for providing secure Skinny Client Control Protocol (SCCP) signaling between Cisco Unified CME and IP phones.	Configuring Security
	Soft keys	**Feature blocking**—The features associated with the following soft keys can be individually blocked per ephone: CFwdAll, Confrn, GpickUp, Park, PickUp, and Trnsfer. The soft key is not removed, but it does not function. **Soft-key control for hold state**—The soft keys that are available while a call is on hold can be modified. The NewCall and Resume soft keys are normally available when a phone has a call on hold, but a template can be applied to the phone to remove these soft keys.	Customizing Soft Keys
	Speed Dial	**Bulk-loading of speed-dial numbers**—Text files with lists of speed-dial numbers can be loaded into system flash or a URL. The files can hold up to 10,000 numbers and can be applied to all ephones or to specific ephones.	Configuring Speed Dial

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	System-Level Parameters	**Disabling automatic phone registration**—Normally, Cisco Unified CME allocates an ephone slot to any ephone that connects to the system. To prevent unauthorized registrations, the **no auto-reg-ephone** command prevents any ephone from registering with Cisco Unified CME if its MAC address is not explicitly listed in the configuration. **External storage of configuration files and per-phone configuration files**—Phone configuration files can be stored on an external TFTP server to offload the TFTP server function of the Cisco Unified CME router. This additional storage space permits the use of per-phone configuration files, which can be used to specify different user locales and network locales for phones. **Failover to Redundant Router**—Sites can be set up with a primary and secondary Cisco Unified CME router to provide redundant Cisco Unified CME capability. Phones automatically register at the secondary router if the primary router fails and later rehome to the primary router when it is operational again.	Configuring System-Level Parameters
	Templates	**Maximum number of ephone templates**—The maximum number of ephone templates that can be defined has increased from 5 to 20. No special configuration is required. **New commands available for ephone templates**—Ephone templates were previously introduced to allow system administrators to control the display of soft keys in various call states on individual ephones. Their role has been expanded to allow you to define a set of ephone parameter values that can be assigned to one or more phones in a single step. **Ephone-dn templates**—Ephone-dn templates are introduced to allow administrators to easily apply sets of configured parameters to individual ephone-dns. Up to 15 ephone-dn templates can be defined.	Creating Templates
	Video Support	**Video support for SCCP-based endpoints**—This feature adds video support to allow you to pass a video stream with a voice call between video-capable SCCP endpoints and between SCCP and H.323 endpoints. Through the Cisco Unified CME router, the video-capable endpoints can communicate with each other locally to a remote H.323 endpoint through a gateway or through an H.323 network.	Configuring Video Support

Table 1-1 Supported Cisco Unified CME Features (continued)

Version	Feature Name	Feature Description	Where Documented
	Voice Mail	**Line-selectable MWI**—Previously, the message-waiting indication (MWI) lamp on a phone could only indicate when messages were waiting for the primary number on a phone. Now, any phone line can be designated during configuration. **Mailbox selection policy for voice-mail servers**—A policy can be set for selecting the mailbox to use for calls that are diverted one or more times within a Cisco Unified CME system before being sent to a Cisco Unity Express, Cisco Unity, or PBX voice-mail pilot number. **Prefix option for SIP unsolicited MWI Notify messages**—Central voice-message servers that provide mailboxes for multiple Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. You can specify the prefix for your site so that central mailbox numbers are correctly converted to your extension numbers.	Integrating Voice Mail
	XML Interface	**XML interface enhancements**—An eXtensible Markup Language (XML) application program interface (API) is provided to supply data from Cisco Unified CME to management software. In Cisco Unified CME 4.0 and later versions, all Cisco Unified CME features have XML support.	Configuring the XML API