



Cisco Unified CME Overview

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Cisco Unified Communications Manager Express (formerly known as Cisco Unified CallManager Express) is a call-processing application in Cisco IOS software that enables Cisco routers to deliver key-system or hybrid PBX functionality for enterprise branch offices or small businesses.

Contents

- [Information About Cisco Unified CME, page 25](#)
- [Where to Go Next, page 32](#)
- [Additional References, page 32](#)
- [Obtaining Documentation, Obtaining Support, and Security Guidelines, page 34](#)

Information About Cisco Unified CME

To design and configure a Cisco Unified Communications Manager Express (Cisco Unified CME) system, you should understand the following concepts:

- [Cisco Unified CME Overview, page 26](#)
- [Licenses, page 27](#)
- [PBX or Keyswitch Model, page 29](#)
- [Call Details Records, page 31](#)
- [Cisco Unified CME on the Cisco 3200 Series, page 31](#)

Cisco Unified CME Overview

Cisco Unified CME is a feature-rich entry-level IP telephony solution that is integrated directly into Cisco IOS software. Cisco Unified CME allows small business customers and autonomous small enterprise branch offices to deploy voice, data, and IP telephony on a single platform for small offices, thereby streamlining operations and lowering network costs.

Cisco Unified CME is ideal for customers who have data connectivity requirements and also have a need for a telephony solution in the same office. Whether offered through a service provider's managed services offering or purchased directly by a corporation, Cisco Unified CME offers most of the core telephony features required in the small office, and also many advanced features not available with traditional telephony solutions. The ability to deliver IP telephony and data routing by using a single converged solution allows customers to optimize their operations and maintenance costs, resulting in a very cost-effective solution that meets office needs.

A Cisco Unified CME system is extremely flexible because it is modular. A Cisco Unified CME system consists of a router that serves as a gateway and one or more VLANs that connect IP phones and phone devices to the router.

Figure 1 shows a typical deployment of Cisco Unified CME with several phones and devices connected to it. The Cisco Unified CME router is connected to the public switched telephone network (PSTN). The router can also connect to a gatekeeper and a RADIUS billing server in the same network.

Figure 1 *Cisco Unified CME for the Small- and Medium-Size Office*

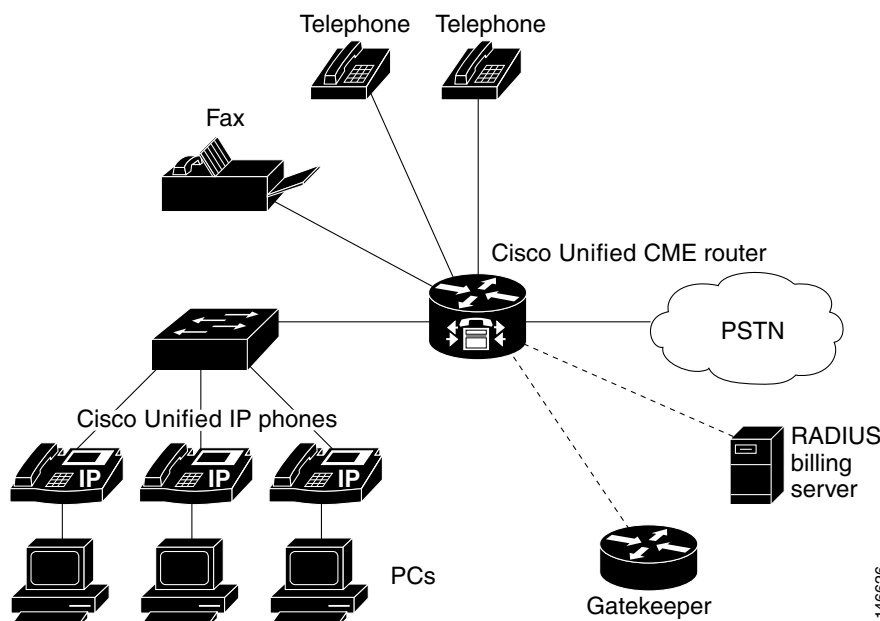
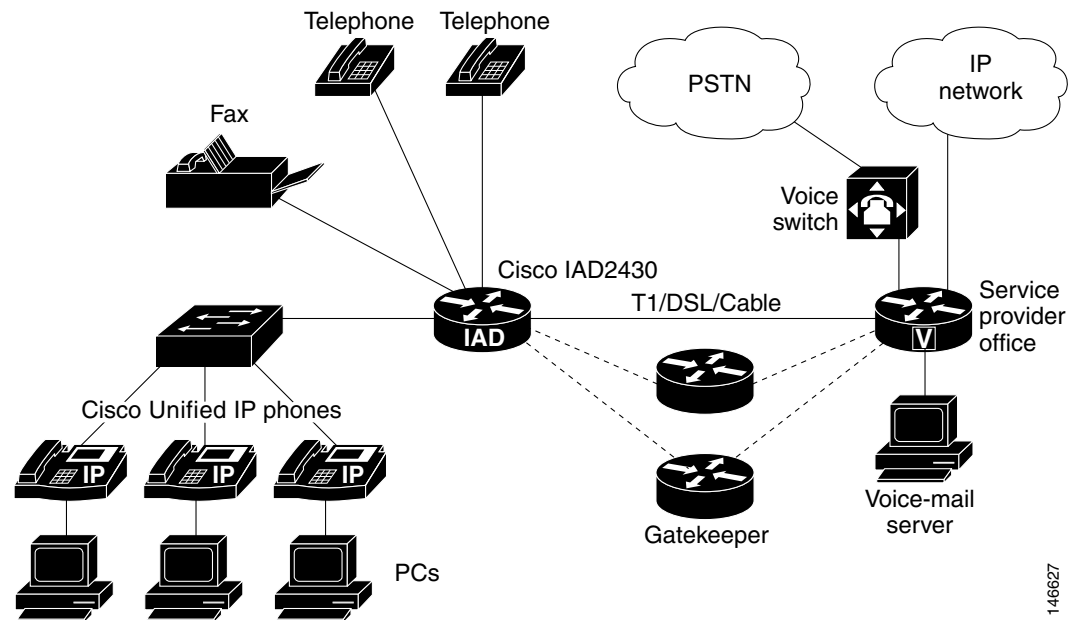


Figure 2 shows a branch office with several Cisco Unified IP phones connected to a Cisco IAD2430 series router with Cisco Unified CME. The Cisco IAD2430 router is connected to a multiservice router at a service provider office, which provides connection to the WAN and PSTN.

Figure 2 Cisco Unified CME for Service Providers



A Cisco Unified CME system uses the following basic building blocks:

- **Ephone or voice register pool**—A software concept that usually represents a physical telephone, although it is also used to represent a port that connects to a voice-mail system, and provides the ability to configure a physical phone using Cisco IOS software. Each phone can have multiple extensions associated with it and a single extension can be assigned to multiple phones. Maximum number of ephones and voice register pools supported in a Cisco Unified CME system is equal to the maximum number of physical phones that can be connected to the system.
- **Directory number**—A software concept that represents the line that connects a voice channel to a phone. A directory number represents a virtual voice port in the Cisco Unified CME system, so the maximum number of directory numbers supported in Cisco Unified CME is the maximum number of simultaneous call connections that can occur. This concept is different from the maximum number of physical lines in a traditional telephony system.

Licenses

You must purchase a base Cisco Unified CME feature license and phone user licenses that entitle you to use Cisco Unified CME. In Cisco Unified CME Release 10, you should purchase:

- [Cisco Unified CME Permanent License](#) or
- [Collaboration Professional Suite License](#)

Cisco Unified CME Permanent License

When you purchase a Cisco Unified CME permanent license, the permanent license is installed on the device when the product is shipped to you. A permanent license never expires and you will gain access to that particular feature set for the lifetime of the device across all IOS release. If you purchase a permanent license for Cisco Unified CME, you do not have to go through the Evaluation Right to Use and Right To Use (RTU) licensing processes for using the features. If you want to purchase a CME-SRST license for your existing device, you have to go through the RTU licensing process for using the features. There is no change in the existing process for purchasing the license.

The Cisco Unified CME permanent license is available in the form of an XML *cme-locked3* file. You should get the XML file and load it in the flash memory of the device. To install the permanent license from the command prompt, use the **license install flash0:cme-locked3** command. The *cme-locked3* is the xml file of the license.

Collaboration Professional Suite License

Collaboration Professional is a new suite of licenses. The Collaboration Professional Suite can be purchased either as a permanent license or an RTU license.

Collaboration Professional Suite Permanent License—When you purchase the Collaboration Professional Suite license, by default, the Cisco Unified CME licenses are delivered as part of the Collaboration Professional Suite. You do not have to separately install and activate the Cisco Unified CME license. The Collaboration Professional Suite permanent license is available in the form of an XML file. You should get the XML file and load it in the flash memory of the device. To install the permanent license from the command prompt, use the **license install flash:lic_name** command.

Collaboration Professional Suite RTU License—When you purchase the Collaboration Professional Suite RTU license, you do not have to go through the Evaluation Right to Use process. However, you have to go through the RTU licensing process for using the Cisco Unified CME features. To install the Collaboration Professional Suite RTU license from the command prompt, use the **license install flash0:colla_pro** command. To activate the license, use the **license boot module c2951 technology-package collabProSuitek9** command.

Restrictions

- For the Cisco Unified CME license, the UCK9 technology package must be available if the Collaboration Professional Suite package is not installed.
- UCK9 is a prerequisite for Cisco Unified CME Release 10.

To activate the Cisco Unified CME feature license, see the [Activating CME-SRST Feature License](#) document.



Note

To support H.323 call transfers and forwards to network devices that do not support the H.450 standard, such as Cisco Unified Communications Manager, a tandem gateway is required in the network. The tandem gateway must be running Cisco IOS Release 12.3(7)T or a later release and requires the Integrated Voice and Video Services feature license (FL-GK-NEW-xxx), which includes the H.323 gatekeeper, IP-to-IP gateway, and H.450 tandem functionality.

PBX or Keyswitch Model

When setting up a Cisco Unified CME system, you need to decide if call handling should be similar to that of a PBX, similar to that of a keyswitch, or a hybrid of both. Cisco Unified CME provides significant flexibility in this area, but you must have a clear understanding of the model that you choose.

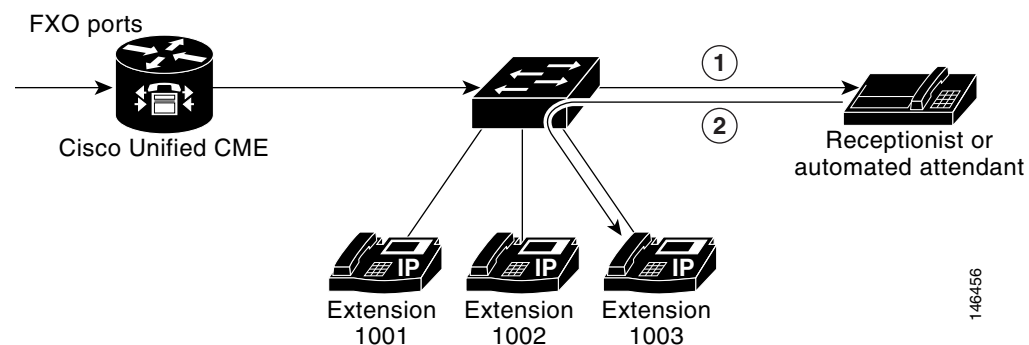
PBX Model

The simplest model is the PBX model, in which most of the IP phones in your system have a single unique extension number. Incoming PSTN calls are routed to a receptionist at an attendant console or to an automated attendant. Phone users may be in separate offices or be geographically separated and therefore often use the telephone to contact each other.

For this model, we recommend that you configure directory numbers as dual-lines so that each button that appears on an IP phone can handle two concurrent calls. The phone user toggles between calls using the blue navigation button on the phone. Dual-line directory numbers enable your configuration to support call waiting, call transfer with consultation, and three-party conferencing (G.711 only).

[Figure 3](#) shows a PSTN call that is received at the Cisco Unified CME router, which sends it to the designated receptionist or automated attendant (1), which then routes it to the requested extension (2).

Figure 3 Incoming Call Using PBX Model



For configuration information, see the [“How to Configure Phones for a PBX System”](#) section on page 226.

Keyswitch Model

In a keyswitch system, you can set up most of your phones to have a nearly identical configuration, in which each phone is able to answer any incoming PSTN call on any line. Phone users are generally close to each other and seldom need to use the telephone to contact each other.

For example, a 3x3 keyswitch system has three PSTN lines shared across three telephones, such that all three PSTN lines appear on each of the three telephones. This permits an incoming call on any PSTN line to be directly answered by any telephone—without the aid of a receptionist, an auto-attendant service, or the use of (expensive) DID lines. Also, the lines act as shared lines—a call can be put on hold on one phone and resumed on another phone without invoking call transfer.

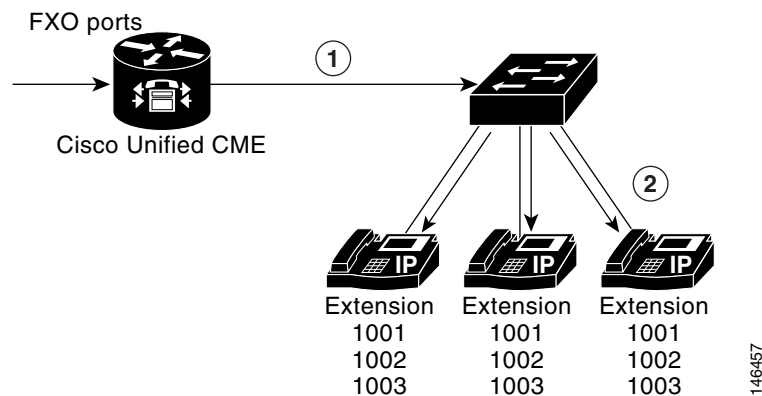
In the keyswitch model, the same directory numbers are assigned to all IP phones. When an incoming call arrives, it rings all available IP phones. When multiple calls are present within the system at the same time, each individual call (ringing or waiting on hold) is visible and can be directly selected by pressing

the corresponding line button on an IP phone. In this model, calls can be moved between phones simply by putting the call on hold at one phone and selecting the call using the line button on another phone. In a keyswitch model, the dual-line option is rarely appropriate because the PSTN lines to which the directory numbers correspond do not themselves support dual-line configuration. Using the dual-line option also makes configuration of call-coverage (hunting) behaviors more complex.

You configure the keyswitch model by creating a set of directory numbers that correspond one-to-one with your PSTN lines. Then you configure your PSTN ports to route incoming calls to those ephone-dns. The maximum number of PSTN lines that you can assign in this model can be limited by the number of available buttons on your IP phones. If so, the overlay option may be useful for extending the number of lines that can be accessed by a phone.

Figure 4 shows an incoming call from the PSTN (1), which is routed to extension 1001 on all three phones (2).

Figure 4 Incoming PSTN Call Using Keyswitch Model



For configuration information, see the [“How to Configure Phones for a Key System”](#) section on page 258.

Hybrid Model

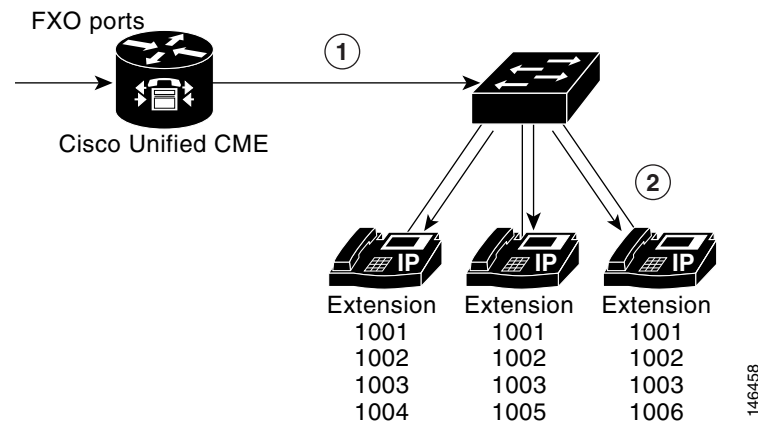
PBX and keyswitch configurations can be mixed on the same IP phone and can include both unique per-phone extensions for PBX-style calling and shared lines for keyswitch-style call operations. Single-line and dual-line directory numbers can be combined on the same phone.

In the simplest keyswitch deployments, individual telephones do not have private extension numbers. Where key system telephones do have individual lines, the lines are sometimes referred to as intercoms rather than as extensions. The term “Intercom” is derived from “internal communication;” there is no assumption of the common “intercom press-to-talk” behavior of auto dial or auto answer in this context, although those options may exist.

For key systems that have individual intercom (extension) lines, PSTN calls can usually be transferred from one key system phone to another using the intercom (extension) line. When Call Transfer is invoked in the context of a connected PSTN line, the outbound consultation call is usually placed from the transferer phone to the transfer-to phone using one of the phone’s intercom (extension) line buttons. When the transferred call is connected to the transfer-to phone and the transfer is committed (the transferer hangs up), the intercom lines on both phones are normally released and the transfer-to call continues in the context of the original PSTN line button (all PSTN lines are directly available on all phones). The transferred call can be put on hold (on the PSTN line button) and then subsequently resumed from another phone that shares that PSTN line.

For example, you can design a 3x3 keyswitch system as shown in [Figure 4](#) and then add another, unique extension on each phone ([Figure 5](#)). This setup will allow each phone to have a “private” line to use to call the other phones or to make outgoing calls.

Figure 5 Incoming PSTN Call Using Hybrid PBX-Keyswitch Model



Call Details Records

The accounting process collects accounting data for each call leg created on the Cisco voice gateway. You can use this information for post-processing activities such as generating billing records and network analysis. Voice gateways capture accounting data in the form of call detail records (CDRs) containing attributes defined by Cisco. The gateway can send CDRs to a RADIUS server, syslog server, or to a file in .csv format for storing to flash or an FTP server. For information about generating CDRs, see [CDR Accounting for Cisco IOS Voice Gateways](#).

Cisco Unified CME on the Cisco 3200 Series

Cisco Unified CME 4.3 and later versions support the Cisco 3200 Series Mobile Access Router.

The Cisco 3200 Series Mobile Access Router offers secure data, voice, and video communications with seamless mobility across wireless networks independent of location or movement. This access router has a high-performance, compact, rugged design optimized for use in vehicles in the defense, public safety, Homeland Security, and transportation markets.

Cisco Unified CME on the Cisco 3200 Series can be deployed in sites requiring on demand network connectivity and voice and data communications that typically do not have PSTN connectivity. The benefits include:

- Ensures voice communications locally if the WAN link fails
- Allows greater autonomy for voice communications at remote sites
- Supports H.323 and SIP trunks
- Easily portable

For information on installing and configuring the Cisco 3200 Series Mobile Access Router, see the [Cisco 3200 Series Mobile Access Router](#) documentation.

Restrictions for Cisco 3200 Series

The Cisco 3200 Series Mobile Access Router has the following restrictions:

- Fixed flash memory
- No PSTN interfaces (FXS, E&M)
- No Advanced Integration Module (AIM)
- No digital signal processor (DSP)
- No analog interfaces
- No T1/E1/BRI digital interfaces
- MGCP is not supported

The following Cisco Unified CME features are not supported on the Cisco 3200 Series:

- SIP Phones
- Basic Automatic Call Distribution (B-ACD) and IVR applications
- Cisco Unified CME GUI
- Cisco Unified Contact Center Express (Unified CCX) integration
- Hardware Conferencing and Transcoding
- Voice Mail

To manage flash memory limitations on the Cisco 3200 Series Mobile Access Router, store phoneloads and other files for Cisco Unified CME on an external TFTP server. Use the **cnf-file location tftp** command as described in the [“SCCP: Defining Per-Phone Configuration Files and Alternate Location” section on page 152](#).

Where to Go Next

Before configuring Cisco Unified CME, see [“Before You Begin” on page 35](#).

Additional References

The following sections provide references related to Cisco Unified CME.

Related Documents

Related Topic	Document Title
Cisco Unified CME configuration	• Cisco Unified CME Command Reference • Cisco Unified CME Documentation Roadmap
Cisco IOS commands	• Cisco IOS Voice Command Reference • Cisco IOS Software Releases 12.4T Command References
Cisco IOS configuration	• Cisco IOS Voice Configuration Library • Cisco IOS Software Releases 12.4T Configuration Guides

Related Topic	Document Title
Cisco IOS voice troubleshooting	• Cisco IOS Voice Troubleshooting and Monitoring Guide
Dial peers, DID, and other dialing issues	• Dial Peer Configuration on Voice Gateway Routers • Understanding One Stage and Two Stage Dialing (technical note) • Understanding How Inbound and Outbound Dial Peers Are Matched on Cisco IOS Platforms (technical note) • Using IOS Translation Rules - Creating Scalable Dial Plans for VoIP Networks (sample configuration)
Dynamic Host Configuration Protocol (DHCP)	• “DHCP” section of the Cisco IOS IP Addressing Services Configuration Guide
Fax and modem configurations	• Cisco Fax Services over IP Application Guide
FXS ports	FXS Ports in H.323 Mode • “Configuring Analog Voice Ports” section of the Cisco IOS Voice Port Configuration Guide • Caller ID FXS Ports in SCCP Mode on Cisco VG 224 Analog Phone Gateway • SCCP Controlled Analog (FXS) Ports with Supplementary Features in Cisco IOS Gateways • Cisco VG 224 Analog Phone Gateway data sheet
H.323	• Cisco IOS H.323 Configuration Guide
Network Time Protocol (NTP)	• “Performing Basic System Management” chapter of Cisco IOS Network Management Configuration Guide
Phone documentation for Cisco Unified CME	• User Documentation for Cisco Unified IP Phones
Public key infrastructure (PKI)	• “Part 5: Implementing and Managing a PKI” in the Cisco IOS Security Configuration Guide
SIP	• Cisco IOS SIP Configuration Guide
TAPI and TSP documentation	• Cisco Unified CME programming Guides
Tcl IVR and VoiceXML	• Cisco IOS Tcl IVR and VoiceXML Application Guide - 12.3(14)T and later • Default Session Application Enhancements • Tcl IVR API Version 2.0 Programmer's Guide • Cisco VoiceXML Programmer's Guide
VLAN class-of-service (COS) marking	• Enterprise QoS Solution Reference Network Design Guide
Voice-mail integration	• Cisco Unified CallManager Express 3.0 Integration Guide for Cisco Unity 4.0 • Integrating Cisco CallManager Express with Cisco Unity Express
Call detail records (CDRs)	• CDR Accounting for Cisco IOS Voice Gateways
XML	• XML Provisioning Guide for Cisco CME/SRST • Cisco IP Phone Services Application Development Notes

Related Websites

Related Topic	Title and Location
Cisco IOS configuration examples	Cisco Systems Technologies website Select a technology category and subsequent hierarchy of subcategories, and then click Configure > Configuration Examples and Tech Notes .

MIBs

MIBs	MIBs Link
CISCO-CCME-MIB MIB CISCO-VOICE-DIAL-CONTROL-MIB	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/techsupport

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>