Cisco Unified CME Overview

- Important Information about Cisco IOS XE 16 Denali, on page 1
- Unified CME Graphical User Interface Deprecation, on page 1
- CTI CSTA Protocol Suite Deprecation, on page 2
- Simple Network Management Protocol (SNMP) Support for Unified CME, on page 3
- Introduction, on page 3
- Licenses, on page 5
- PBX or Keyswitch, on page 8
- Call Detail Records, on page 10
- Additional References, on page 10

Important Information about Cisco IOS XE 16 Denali

Effective Cisco IOS XE Release 3.7.0E (for Catalyst Switching) and Cisco IOS XE Release 3.17S (for Access and Edge Routing) the two releases evolve (merge) into a single version of converged release—the Cisco IOS XE 16 Denali—providing one release covering the extensive range of access and edge products in the Switching and Routing portfolio.

For migration information related to the Cisco IOS XE 16, see Cisco IOS XE Denali 16.2 Migration Guide for Access and Edge Routers.

Unified CME Graphical User Interface Deprecation

From Unified CME Release 12.6 (Cisco IOS XE Gibraltar 16.11.1a Release), the Graphical User Interface (GUI) is no more supported for Unified CME. Hence, the GUI files posted under the name cme-gui-..., as part of the Unified CME software bundle, is not available for download for Unified CME 12.6 and later releases. We recommend that you use the Command Line Interface (CLI) commands to configure Unified CME.

All the CLI commands related to Unified CME GUI deployment are disabled for Unified CME 12.6 and later releases. The following CLI commands related to Unified CME GUI are disabled:

- `web admin customer name username {password string|secret {0|5} string}
- `web admin system [name username] [{password string|secret {0|5} string}]
- `web customize load filename
CTI CSTA Protocol Suite Deprecation

From Unified CME Release 12.6 (Cisco IOS XE Gibraltar 16.11.1a), the Computer Telephony Integration (CTI) Computer Supported Telecommunications Applications (CSTA) protocol suite is no more supported on Unified CME. All the CLI commands related to CTI CSTA are disabled for Unified CME 12.6 and later releases.

The following CLI commands related to CTI CSTA that are configured under \texttt{voice service voip} configuration mode is disabled on Unified CME 12.6 and later releases:

- \texttt{cti shutdown}
- \texttt{cti callmonitor}
- \texttt{cti csta mode basic}
- \texttt{cti message device-id suppress-conversion}
- \texttt{cti timeout make-call-prompt}

The following CLI commands related to CTI CSTA that are configured under \texttt{ephone-dn} and \texttt{ephone-template} configuration mode is disabled on Unified CME 12.6 and later releases:

- \texttt{cti notify}
- \texttt{cti watch}

The following CLI commands related to CTI CSTA that are configured under \texttt{voice register session-server} configuration mode are disabled on Unified CME 12.6 and later releases:

- \texttt{cti aware}

The following CLI show commands related to CTI CSTA that are configured under \texttt{show cti} are disabled on Unified CME 12.6 and later releases:

- \texttt{show cti call}
- \texttt{show cti gcid}
- \texttt{show cti line-node}
- \texttt{show cti session}
Simple Network Management Protocol (SNMP) Support for Unified CME

Unified CME supports Simple Network Management Protocol (SNMP) Management Information Base (MIBs) for monitoring the product status. Unified CME Release 12.6 and later is SNMP Version 3 (SNMPv3) compliant. Unified CME supports the following main SNMP MIB:

- CISCO-CCME-MIB

For information on configuration of SNMP version 3 on Unified CME router, see SNMP Configuration Guide.

Introduction

The Cisco Unified Communications Manager Express System Administrator Guide refers to a phone with SIP firmware as SIP Phone, SIP IP Phone, or Cisco Unified SIP IP phone. A phone with SCCP firmware is referred as SCCP Phone, SCCP IP Phone, or Cisco Unified SCCP IP phone.

Note

It is mandatory to configure the command supplementary-service media-renegotiate under voice service voip configuration mode to enable the supplementary features supported on Unified CME.

Note

Configure the CLI commands no supplementary-service sip refer, no supplementary-service sip moved-temporarily under voice service voip configuration mode for call transfer and call forward scenarios in Unified CME.

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.

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: Cisco Unified CME for the Small- and Medium-Size Office, on page 4 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 2: Cisco Unified CME for Service Providers, on page 4 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.

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 11, you should purchase:

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.

Cisco Smart License

From Release 11.7 onwards, Cisco Unified CME supports Smart Licensing, apart from the existing CSL licensing model. Smart Licensing is supported only on Cisco 4000 Series Integrated Services Router. Depending on the technology package available on the router, licenses such as UCK9 and Security are supported using Smart Licensing.

The Smart Licensing feature is a software based licensing model that gives you visibility into license ownership and consumption. The Smart Licensing model consists of a web interface named Cisco Smart Software Manager (CSSM). CSSM is a central license repository that manages licenses across all Cisco products that you own, including Unified CME. Your access to the CSSM account is authenticated using valid Cisco credentials.

You can use the CSSM Smart Account to generate valid tokens IDs. The token IDs are used to register the Unified CME device with your CSSM Smart Account. Once the token is generated, it can be used to register many other product instances in your network.
On the Unified CME router, you need to ensure that the call home feature is not disabled. Also, Smart Licensing feature should be enabled at the router using the CLI command `license smart enable`. Use the `no` form of the command to disable smart licensing. For more information on configuring Smart Licensing in your router, see Software Activation Configuration Guide, Cisco IOS Release 15M&T. Once the smart license is enabled and the router is not yet registered with CSSM, the device enters an evaluation period of 90 days.

You can register the router to CSSM with the token ID. To register the device (Unified CME router) with CSSM, use the CLI command `license smart register idtoken`. For information on registering the device with CSSM, see Software Activation Configuration Guide.

Upon successful registration, Unified CME is in Registered status. Then, Unified CME sends an authorization request for all the phones configured. Based on the licenses in the Smart Account, CSSM responds with one of the defined statuses such as Authorized (using less than it has licenses for) or Out-of-Compliance (using more than it has licenses for).

The CSSM assigns licenses that are available in your CSSM account to the phones configured across the routers. Unified CME supports only one license entitlement to validate phones configured on Unified CME.

- **CME_EP**—This license type supports all phones configured on Unified CME.

---

**Note**

The CME_EP license count reflects the total phone count of both the ephones and pools that are configured in the Unified CME irrespective of whether the phones are registered or not.

Unified CME sends an authorization request when a license consumption changes or every 30 days to let CSSM know it's still available and communicating. The ID certificate issued to identify Unified CME at time of registration is valid for one year, and is automatically renewed every six months.

---

**Note**

If the router does not communicate with CSSM for a period of 90 days, the license authorization expires.

The license count is evaluated for the number of phones configured across the routers. The CSSM Licenses page reflects the total license count usage. The total number of licenses available for a type of license (Quantity), number of licenses currently use (In Use), and the number of unused or over-used licenses (Surplus/Shortage).

For example, consider a smart account in CSSM with 50 CME_EP licenses. If the user has a single registered Unified CME with 20 analog phones configured, the CSSM licenses page will reflect Quantity as 50, In Use as 20, and Surplus as 30. For more information on Smart Software manager, see Cisco Smart Software Manager User Guide.

Once a new phone is configured on a Unified CME registered with CSSM, a timer is initiated to report the phone configuration to CSSM. A new phone configuration is reported only at the end of the time set in the timer (3 minutes). Hence, the Smart Agent reports all the new configurations created within the time period defined using the preset timer. The CSSM increments the license usage count based on the report sent by Smart Agent. If the number of phones configured exceeds the license limit, then CSSM generates an alert in the user account. When a phone configuration is removed, the license usage count is decremented.

The license entitlement for Unified CME smart license is displayed on the router as follows:

```
Router# show license summary
Smart Licensing is ENABLED

Registration:
```
Status: REGISTERED
Smart Account: Call-Manager-Express
Virtual Account: CME Application
Export-Controlled Functionality: Not Allowed
Last Renewal Attempt: None
Next Renewal Attempt: Oct 07 12:08:10 2016 UTC

License Authorization:
Status: AUTHORIZED
Last Communication Attempt: SUCCESS
Next Communication Attempt: May 13 07:11:48 2016 UTC

License Usage:

<table>
<thead>
<tr>
<th>License</th>
<th>Entitlement tag</th>
<th>Count</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>regid.2014-12.com.ci...</td>
<td>(ISR_4351_UnifiedCommun..)</td>
<td>1</td>
<td>AUTHORIZED</td>
</tr>
<tr>
<td>regid.2016-10.com.ci...</td>
<td>(CME_EP)</td>
<td>4</td>
<td>AUTHORIZED</td>
</tr>
</tbody>
</table>

Licensing Modes

From Unified CME 11.7 onwards, both CSL and Smart Licensing modes are supported. That is, customers can continue with CSL by not enabling Smart Licensing. Alternatively, they can enable Smart Licensing and decide later to go back to CSL by disabling Smart Licensing with the `no license smart enable` command. When you switch to CSL from the Smart Licensing mode, you need to ensure that the End User License Agreement (EULA) is signed. CSL is not supported unless the EULA is signed. Use the CLI command `license accept end user agreement` in global configuration mode to configure EULA.

To verify the status of the license issued to phones registered on Unified CME, you can use the `show license` command.

Router#show license ?
  all         Show license all information
  status      Show license status information
  suites      Show license suite information
  summary     Show license summary
  tech        Show license tech support information
  udi         Show license udi information
  usage       Show license usage information

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 11.

Note

As compared to Unified CME Release 10.5 and prior, all the future releases of Unified CME displays the CME-SRST license state as Active, Not in Use. This is applicable when Unified CME is removed from the router (configure `no telephony-service` and `no voice register global` to remove Unified CME from the router).

To activate the Cisco Unified CME feature license, see Activating CME-SRST Feature License.
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.

Note

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: Incoming Call Using PBX Model, on page 8 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).

For configuration information, see Configure Phones for a PBX System.

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: Incoming PSTN Call Using Keyswitch Model, on page 9 shows an incoming call from the PSTN (1), which is routed to extension 1001 on all three phones (2).

For configuration information, see Configure Phones for a Key System.

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 transferrer 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 transferrer 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: Incoming PSTN Call Using Keyswitch Model, on page 9 and then add another, unique extension on each phone (Figure 5: Incoming PSTN Call Using Hybrid PBX-Keyswitch Model, on page 10). 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](image)

**Call Detail 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.

**Additional References**

The following section provides references related to Cisco Unified CME.

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified CME</td>
<td>Cisco Unified CME Command Reference</td>
</tr>
<tr>
<td>configuration</td>
<td>Cisco Unified CME Documentation Roadmap</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco IOS commands</td>
<td>Cisco IOS Voice Command Reference</td>
</tr>
<tr>
<td></td>
<td>Cisco IOS Software Releases 12.4T Command References</td>
</tr>
<tr>
<td>Related Topic</td>
<td>Document Title</td>
</tr>
<tr>
<td>---------------</td>
<td>----------------</td>
</tr>
</tbody>
</table>
| Cisco IOS configuration | Cisco IOS Voice Configuration Library  
Cisco IOS Software Releases 12.4T Configuration Guides |
| 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 SCCP Mode on Cisco VG 224 Analog Phone Gateway  
“Configuring Analog Voice Ports” section of the Cisco IOS Voice Port Configuration Guide  
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  
Cisco Voice XML Programmer’s Guide |
| VLAN class-of-service (COS) marking | Enterprise QoS Solution Reference Network Design Guide |
### Management Information Base

<table>
<thead>
<tr>
<th>MIBs</th>
<th>MIBs Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>CISCO-CCME-MIB</td>
<td>To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: <a href="http://www.cisco.com/go/mibs">http://www.cisco.com/go/mibs</a></td>
</tr>
<tr>
<td>MIB</td>
<td></td>
</tr>
<tr>
<td>CISCO-VOICE-DIAL-CONTROL-MIB</td>
<td></td>
</tr>
</tbody>
</table>