Cisco Systems

Cisco Unified CallManager Express Configuration Guide for SIP Phones

June 19, 2006

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

Text Part Number: OL-10782-01

CONTENTS

Overview of SIP Phones in Cisco Unified CME

Revised: June 19, 2006

Note The name of this product was changed to Cisco Unified CallManager Express in version 4.0.

Cisco CME 3.4 added station-side RFC 3261 standard-based support for Session Initiation Protocol (SIP) phones directly into Cisco Unified CallManager Express (Cisco Unified CME). This enables Cisco Unified IP phones to place calls across SIP networks in the same way that the current Skinny Client Control Protocol (SCCP) phones do.

Finding Feature Information

Your Cisco IOS software release may not support all of the features documented in this guide. To reach links to specific feature documentation in this guide and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Unified CME for SIP Phones" section in this guide.

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at
http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Finding Support Information for Cisco Unified CME

For information about Cisco IOS software and Cisco Unified CME compatibility, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

For information about Cisco Unified CME specifications, including number of supported phones, see the appropriate *Cisco Unified CME Firmware, Platforms, Memory, and Voice Products* document at
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Contents

Software License

Each SIP phone requires a SIP user license. The following SIP user licenses support Cisco Unified IP phones:

- SW-SM-UL-79XX for Cisco Unified IP Phone 7960s and Cisco Unified IP Phone 7940s
- SW-SMH-UL-79XX for Cisco Unified IP Phone 7905s and Cisco Unified IP Phone 7912s
- SW-SMH-UL-ATA-1P for ATA; one license per port
- Existing SCCP user license can be used to support a SIP phone

Cisco Unified CME requires an FL-CCME-XX license. The following limitations apply to the FL-CCME-XX license:

- Each SIP phone will consume one set of license
- The total number of SIP + SCCP phones connected to a Cisco Unified CME cannot exceed the maximum allowed by the FL-CCME license. For example, FL-CCME-36 can be used to support 20 SCCP phones + 16 SIP phones or 36 SIP phones

Prerequisites for Configuring Cisco Unified CME for SIP Phones

The following prerequisites apply to configuring Cisco Unified CME support for SIP phones:

- Install Cisco Unified CME. To find installation instructions, se the documenatation at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.
- Set up your Cisco Unified CME system. To find configuration tasks for setting up Cisco Unified CME, see the *Cisco Unified CallManager Express System Administrator Guide* for your version at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.
- To provide voice-mail support for SIP phones connected to Cisco Unified CME, install and configure voice mail on your network.

Restrictions for Configuring Cisco Unified CME for SIP Phones

All restrictions described in the *Cisco Unified CallManager Express System Administrator Guide* for your version apply to SIP phone support; see http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Information About Cisco Unified CME Support for SIP Phones

Note For more information about Cisco Unified CME, see the Cisco Unified CallManager Express: All Versions web page—including the *Cisco Unified CallManager Express Specifications*, *Cisco Unified CallManager Express System Administrator Guide*, the *Cisco Unified CallManager Express Command Reference*, and other documents—at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Station-side RFC 3261 standard-based support for Session Initiation Protocol (SIP) phones directly into Cisco Unified CME is in Cisco CME 3.4 and later. This enables Cisco Unified IP phones to place calls across SIP networks in the same way that the current Skinny Client Control Protocol (SCCP) phones do. Cisco Unified CME acts as the primary SIP registrar server for SIP phones.

The following call combinations are supported in Cisco Unified CME:

- Direct calls between local SIP phones
- Incoming and outgoing calling between SIP phones
- Incoming and outgoing calling between a SIP phone and a SCCP phone
- SIP phone to WAN VoIP using SIP protocol

Cisco Unified CME supports Media Flow-through mode only; enabling SIP-to-SIP calls is required before you can successfully make SIP-to-SIP calls. The following sample shows the required configuration:

```
voice service voip
 allow-connection sip to sip
```

All signaling and media between SIP endpoints must flow through Cisco Unified CME. For SCCP phones, the media bypasses Cisco Unified CME for local phone-to-phone calls (flow-around). Both SIP and SCCP phones send VoIP media and signaling through Cisco Unified CME.

For a list of supported features for SIP phones in Cisco Unified CME, see the "Feature Information for Cisco Unified CME for SIP Phones" section.

Cisco IOS Software Configuration Modes

You can provision both SIP and SCCP phones on your Cisco Unified CME router by using the same software command-line interface (CLI). New and modified commands for configuring SIP phones are similar but not identical to the commands for configuring support for SCCP phones. To support phone provisioning for SIP phones, a subset of the voice register pool configuration mode is modified to accept per-phone configuration parameters. Commands under the following configuration modes are used to configure Cisco SIP IP phones on your Cisco Unified CME router:

- Voice register global to set global parameters for all supported SIP phones in Cisco Unified CME.
- Voice register pool to set up phone-specific parameters for extensions that appear on a SIP phone in Cisco Unified CME.
- Voice register dn to define an extension for a SIP phone line, intercom line, voice-mail port, or a message-waiting indicator (MWI).
- Voice register template to define templates of common parameters for assigning phone features to SIP phones.
- Voice hunt-group to apply the Hunt Group feature to a list of SIP phone extensions for redirecting calls for a specific number (hunt-group pilot number) to a defined group of voice directory numbers.

Table 1 illustrates the correlation between ephone and voice register configuration modes.

Table 1 Configuration Modes

Configuration Mode for SIP Phones	Equivalent Configuration Mode for SCCP Phones
voice register global	telephony-service
voice register pool	ephone
voice register dn	ephone-dn
voice register template	ephone-template
voice hunt-group	ephone-hunt group

Note This guide includes information for configuring SIP phones in Cisco Unified CME only and does not include procedures for configuring SIP in Cisco SRST. To find information about configuring SIP in Cisco SRST, see the *Cisco Unified SIP SRST System Administrator Guide* at http://www.cisco.com/en/US/products/sw/voicesw/ps2169/products_documentation_roadmap09186a008018912f.html.

If you work in the Cisco Unified CME environment and would like to know which commands are also applicable to SIP in Cisco Unified SRST, Table 1 is a list of new or modified commands for Cisco CME 3.4 and Cisco SRST 3.4. Versions of the commands marked under the column "Applicable in Cisco CME Only" are available only after the **mode** *cme* command is configured under the voice register global configuration mode. When used with **mode** *cme*, these commands apply to provisioning SIP phones in Cisco Unified CME.

Table 2 New or Modified Commands in Cisco CME 3.4—Sorted by Function and Configuration Mode

Function—> Command	Phone Profile	Dial Peer	Voice Register Mode	Configurable for Cisco CME and Cisco SRST	Applicable to Cisco CME Only
after-hour	—	X	dn	X	—
call forward	*Depends on phone type	X	dn	X	—
huntstop	—	X	dn	X	—
number	X	X	dn	X	—
preference	—	X	dn	X	—

Table 2 New or Modified Commands in Cisco CME 3.4—Sorted by Function and Configuration Mode (continued)

Function—> Command	Phone Profile	Dial Peer	Voice Register Mode	Configurable for Cisco CME and Cisco SRST	Applicable to Cisco CME Only
auto-answer	X	—	dn	—	X
label	X	—	dn	—	X
mwi			dn	—	X
name	X	—	dn	—	X
no-reg	—	X	dn	X	
application	—	X	global	X	—
external ring	X	—	global	X	—
max-dn	—	—	global	X	—
max-pool	—	—	global	X	—
mode	—	—	global	X	—
bulk	—	X	global	—	X
authenticate	—	—	global	—	X
create	—	—	global	—	X
date-format	X	—	global	—	X
dst	X	—	global	—	X
hold-alert	X	—	global	—	X
file	—	—	global	—	X
load	X	—	global	—	X
logo	X	—	global	—	X
mwi	X	—	global	—	X
phone-redirect-limit	X	—	global	—	X
reset	—	—	global	—	X
tftp-path	—	—	global	—	X
time-format	—	X	global	—	X
timezone	X	—	global	—	X
upgrade	X	—	global	—	X
url	X	—	global	—	X
voicemail	X	—	global	—	X
after hour exempt	—	X	pool	X	—
alias			pool	X	
application	—	X	pool	X	—
call-forward	X	—	pool	X	—
codec	X	X	pool	X	—
dtmf-relay	—	X	pool	X	—
id	X	—	pool	X	—

Table 2 New or Modified Commands in Cisco CME 3.4—Sorted by Function and Configuration Mode (continued)

Function—> Command	Phone Profile	Dial Peer	Voice Register Mode	Configurable for Cisco CME and Cisco SRST	Applicable to Cisco CME Only
number	X	X	pool	X	—-
preference	—	X	pool	X	—
proxy	X	X	pool	X	—
translate-outgoing	—	X	pool	X	—
vad	X	X	pool	X	—
call-waiting	X	—	pool	—	X
description	X	—	pool	—	X
dnd	X	—	pool	—	X
forwarding	X	—	pool	—	X
keep-conference	X	—	pool	—	X
reset	—	—	pool	—	X
template	X	—	pool	—	X
type	—	—	pool	—	X
username	X	—	pool	—	X
anonymous	X	—	template	—	X
conference	X	—	template	—	X
caller-id	X	—	template	—	X
dnd-control	X	—	template	—	X
speed-dial	X	—	template	—	X
transfer	X	—	template	—	X

Note For complete descriptions of all new and modified commands for Cisco Unified CME, see the the *Cisco Unified CallManager Express Command Reference* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

How SIP Phones Register

Cisco Unified CME acts as the primary registrar server for your SIP phones.

Phones request service from Cisco Unified CME via the IP address specified by the **source-address (voice register global)** command. The registrar confirms the username, i.e. the phone number for the phone. The phone accesses the configuration profile for the individual SIP phone on the TFTP server and processes the information contained in the file, registers itself, and puts the phone number on the phone console display.

Minimally, a SIP configuration profile contains the MAC address and type for a particular Cisco SIP IP phone. Individual SIP phones that are permitted to send and handle Registration messages are identified using the commands in voice register pool configuration mode. Configure the **id** command first to identify a particular individual SIP phone. Use the **number** command to specify the phone number that is permitted by the registrar to handle the Register message from the SIP phone that was identified by using the **id** command. Phone type is either automatically detected by the Cisco CME router or is specified using the **type** command. The **type** command is required only for Cisco ATA phones.

Even though there are similarities in contents for all phone types, file names, format, and number of files are different for various phones. Profiles created for each SIP phone and written to the TFTP server are identified by file aliases based on phone type. Use the **create profile (voice register global)** command to create the following configuration profiles for Cisco SIP IP phones.

- Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7960

  SIPDefault.cnf—Shared among Cisco Unified Phone 7940s and Cisco Unified Phone 7960s

  SIP<mac-address>.cnf—Per phone

  syncinfo.xml—Shared among Cisco Unified IP phone 7940s and Cisco Unified Phone 7960s
- Cisco Unified IP Phone 7905

  Id<mac-address>—Per phone

  Id<mac-address>.txt— (Optional) Created by using the **text file** command.
- Cisco Unified IP phone 7912

  gk<mac-address>—Per phone

  gk<mac-address>.txt— (Optional) Created by using the **text file** command.
- Cisco ATA-186 and Cisco ATA-188

  ata<mac-address>—Per phone

  ata<mac-address>.txt— (Optional) Created by using the **text file** command.

Any time you create or modify parameters under the voice register dn or voice register pool configuration modes, use the **create profile** command to generate a new configuration profile and properly propagate the parameters.

The Cisco universal application loader for phone firmware files allows you to add additional phone features across all protocols, including SIP and SCCP. To do this, a hunt algorithm searches for multiple configuration files. Depending on which *matching* configuration file is found first, the phone automatically selects that protocol. To ensure that Cisco Unified IP phones download the appropriate configuration files for the desired protocol, SCCP or SIP, you must properly configure the SIP phones *before* connecting and rebooting the phones. The hunt algorithm searches for files in the following order:

1. CTLSEP<mac> file for SCCP phones—For example, CTLSEP003094C25D2E.tlv
2. SEP <mac> file for SCCP phones—For example, SEP003094C25D2E.cnf.xml
3. SIP <mac> file for SIP phones—For example, SIP003094C25D2E.cnf or gk003069C25D2E
4. XML default file for SCCP phones—For example, SEPDefault.cnf.xmls
5. SIP default file for SIP phones—For example, SIPDefault.cnf

The configuration profiles are downloaded when a phone is rebooted or reset. The profiles contain the image_version parameter that tells the phone which phone firmware files it should be running. A phone firmware file is associated with the specific Cisco IP phone type by using the **load (voice register global)** command.

Cisco Unified IP phone platforms support both the SCCP and SIP protocols. When Cisco IP phone downloads the configuration profile, the phone compares the firmware specified in the file with the firmware already installed on the phone. If the firmware version differs from the one that is currently loaded on the phone, the phone contacts the TFTP server to upgrade to the new image. The phone downloads the new firmware before registering in Cisco Unified CME. The phones goes through the boot-up sequence again after downloading the new firmware.

In order to support the SIP protocol, brand new, out-of-the-box Cisco Unified IP phones and all already-configured SCCP phones that you now want to use SIP must download Cisco SIP phone firmware files before they can register.

Additional References

The following sections provide references related to Cisco Unified CME.

Related Documents

Related Topic	Document Title
Additional Cisco IOS Voice Configuration Library documents, including library preface and glossary	Cisco IOS Voice Configuration Library at http://www.cisco.com/en/US/products/sw/iosswrel/ps5187/products_configuration_guide_chapter09186a008045dca4.html
Cisco IOS command references	• *Cisco IOS Debug Command Reference,* Release 12.4 at http://www.cisco.com/en/US/products/ps6350/products_command_reference_book09186a008042deb2.html • *Cisco IOS Voice Command Reference,* Release 12.3T at http://cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html
Cisco IOS configuration examples	Cisco Systems Technologies website at http://cisco.com/en/US/tech/index.html **Note** From the website, select a technology category and subsequent hierarchy of subcategories, then click **Technical Documentation > Configuration Examples**.
Cisco IOS troubleshooting information	*Cisco IOS Voice Troubleshooting and Monitoring Guide* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/voipt_c/index.htm
Cisco Unified CME command reference	*Cisco Unified CallManager Express Command Reference* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html
Cisco Unified CME specifications, system administrator guide, Cisco Unified IP phone (Analog and SIP) documentation, and other documents	Cisco Unified Callmanager Express: All Versions website at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html

Standards

Standards	Title
No new or modified standards are supported by this feature, and support for existing standards has not bee modified by this feature.	—

MIBs

MIBs	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not bee modified by this feature	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

RFCs

RFCs	Title
RFC 2543	*SIP: Session Initiation Protocol*
RFC 3261	*SIP: Session Initiation Protocol*

Technical Assistance

Description	Link
The Cisco Technical Support & Documentation website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Upgrading, Downgrading, and Converting Cisco Phone Firmware Files

Revised: August 2, 2006

This chapter describes how to configure the **upgrade** command to ensure that the correct Cisco phone firmware is downloaded by SIP phones in Cisco Unified CallManager Express (Cisco Unified CME).

During registration, a hunt algorithm is employed to search for the configuration file for the particular IP phone that is attempting to register. Depending on which *matching* configuration file is found first, the phone automatically selects that protocol. To ensure that IP phones download the appropriate configuration files for the desired protocol, SCCP or SIP, you must properly configure the IP phones *before* connecting and rebooting the phones. The hunt algorithm searches for files in the following order:

1. CTLSEP<MAC address> file for SCCP phones—For example, CTLSEP000A8A2C8C6E.tlv
2. SEP <MAC address> file for SCCP phones—For example, SEP000A8A2C8C6E.cnf.xml
3. SIP <MAC address> file for SIP phones—For example, SIP000A8A2C8C6E.cnf or gk003069C25D2E
4. XML default file for SCCP phones—For example, SEPDefault.cnf.xmls
5. SIP default file for SIP phones—For example, SIPDefault.cnf

Cisco Unified IP phone platforms support both SCCP and SIP protocols. When a SIP phone downloads its configuration profile, the phone compares the phone firmware mentioned in the configuration profile with the firmware already installed on the phone. If the firmware version differs from the one that is currently loaded on the phone, the phone contacts the TFTP server to upgrade to the new phone firmware. The phone downloads the new firmware before registering with Cisco Unified CME. In order to support the SIP protocol, brand new, out-of-the-box Cisco Unified IP phones and all already-configured SCCP phones that you now want to use SIP, the Cisco Unified IP phone must download the SIP phone firmware before the phone can register. After downloading the new firmware, the phone goes through the boot up sequence again and registers with Cisco Unified CME.

Early versions of Cisco phone firmware for SCCP and SIP IP phones had filenames as follows:

- SCCP firmware—P003xxyy.bin
- SIP firmware—P0S3xxyy.bin

In both bases, x represents the major version, and y represented the minor version. The third character represents the protocol, “0” for SCCP or “S” for SIP.

In current versions, the following conventions are used:

- SCCP firmware—P003xxyyzzww.bin or SCCPxxyyzzww.bin, where x represents the major version, y represents the major subversion, z represents the maintenance version, and w represents the maintenance subversion.

- SIP firmware—P0S3-xx-y-zz, where x represents the major version, y represents the minor version, and z represents the subversions.
- The third character in the file name—Represents the protocol, “0” for SCCP or “S” for SIP.

Table 3 contains firmware-naming convention examples:

Table 3 Firmware-Naming Conventions.

SCCP Phones		SIP Phones	
Image	**Version**	**Image**	**Version**
P00303030300	3.3(3)	P0S3-04-4-00	4.4
P00305000200	5.0(2)	P0S3-05-2-00	5.2
P00306000100	6.0(1)	P0S3-06-0-00	6.0
SCCP08000300	8.0(3)	P0S3-07-1-00	7.1

Caution If your Cisco Unified CME system supports SCCP as well as SIP phones, do not connect your SIP phones to the network until after you have verified the phone configuration profiles.

Contents

Prerequisites for Upgrading, Downgrading, and Converting Cisco Phone Firmware Files

SCCP and SIP phone firmware for Cisco Unified IP phones to be connected to Cisco Unified CME, including all versions required during an upgrade or downgrade sequence, must be loaded in the flash memory of the TFTP server from which the SIP phones will download their configuration profiles.

Tip

- Firmware files can be downloaded individually from http://www.cisco.com/pcgi-bin/tablebuild.pl/ip-iostsp, or from a compressed archive, such as cme-xxx.zip or cme-basic-xxx.tar.
- For a list of the firmware files for each phone type, see the *Cisco Unified CME Supported Firmware, Platforms, Memory, and Voice Products* documentation for your version at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.
- Install only the firmware files for the types of phones that you have at your site.

Restrictions for Upgrading, Downgrading, and Converting Cisco Phone Firmware Files

The manufacturing firmware that comes installed on new Cisco Unified IP Phone 7960s and 7960Gs and Cisco Unified IP Phone 7940s and 7940Gs limits the size of the file that the phone can download to 384K. The phone will fail to upgrade if the file size to be downloaded is larger than 384K. This places a restriction only on the first new file that is downloaded to the Cisco Unified IP Phone 7960 and Cisco Unified IP Phone 7940. In general, after the phone has accepted its first download and replaced the firmware installed by manufacturing, additional upgrades or changes do not have this limitation.

Upgrading or Downgrading Cisco Phone Firmware for SIP Phones Between Versions

To upgrade or downgrade SIP phone firmware for Cisco Unified IP phones between release versions, perform the steps in this section.

The upgrade and downgrade sequences for SIP phones differ per phone type as follows:

- Upgrading/downgrading the phone firmware for Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7912G, and Cisco ATA Analog Telephone Adapter is straightforward; modify the **load** command to upgrade directly to the target load.
- The phone firmware version upgrade sequence for Cisco Unified IP Phone 7940Gs and 7960Gs is from version [234].x to 4.4, to 5.3, to 6.x, to 7.x. You cannot go directly from version [234].x to version 7.x.
- To downgrade phone firmware for Cisco Unified IP Phone 7940Gs and 7960Gs, first upgrade to version 7.x, then modify the **load** command to downgrade directly to the target phone firmware.

Restrictions

- Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7912G, and Cisco ATA—Signed load starts from SIP v1.1. After you upgrade the firmware to a signed load, you cannot downgrade the firmware to an unsigned load.

- Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G—Signed load starts from SIP v5.x. Once you upgrade the firmware to a signed load, you cannot downgrade the firmware to an unsigned load.
- The procedures for upgrading phone firmware files for SIP phones is the same for all Cisco Unified IP phones. For other limits on firmware upgrade between versions, see the phone firmware upgrade matrix at: http://www.cisco.com/en/US/products/sw/voicesw/ps4967/prod_installation_guides_list.html.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **load** *phone-type firmware-file*
6. **upgrade**
7. Repeat Steps 5 and 6.
8. **file text**
9. **create profile**
10. **exit**
11. **voice register pool** *tag*
12. **reset**
13. **exit**
14. **voice register global**
15. **no upgrade**
16. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**load** *phone-type firmware-file* **Example:** `Router(config-register-global)# load 7960-7940 P0S3-06-0-00`	Associates a phone type with a phone firmware file. • A separate **load** command is required for each IP phone type. • *phone-type*—Type **?** to display phone types or see the **load** command in the *Cisco Unified CME Command Reference*. • *firmware-file*—Filename to be associated with the specified Cisco Unified IP phone type. For Cisco ATA only, use the .sbin file extension. For all other Cisco Unified IP phone types, do not use the .sbin file extension.
Step 6	**upgrade** **Example:** `Router(config-register-global)# upgrade`	Generates a file with the universal application loader image for upgrading phone firmware and performs the TFTP server alias binding.
Step 7	Repeat previous two steps. **Example:** `Router(config-register-global)# load 7960-7940 P0S3-07-4-00` `Router(config-register-global)# upgrade`	(Optional) Repeat for each version required in multistep upgrade sequences only.
Step 8	**file text** **Example:** `Router(config-register-global)# file text`	(Optional) Generates ASCII text files for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186, or Cisco ATA-188.
Step 9	**create profile** **Example:** `Router(config-register-global;)# create profile`	Generates provisioning files required for SIP phones and writes the file to the location specified with the **tftp-path** command.
Step 10	**exit** **Example:** `Router(config-register-global)# exit`	Exits from the current command mode to the next highest mode in the configuration mode hierarchy.

	Command or Action	Purpose
Step 11	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 12	**reset** **Example:** `Router(config-register-pool)# reset`	Performs a complete reboot of the single SIP phone specified with the **voice register pool** command and contacts the DHCP server and the TFTP server for updated information.
Step 13	**exit** **Example:** `Router(config-register-pool)# exit`	Exits from the current command mode to the next highest mode in the configuration mode hierarchy.
Step 14	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 15	**no upgrade** **Example:** `Router(config-register-global)# no upgrade`	Return to the default for the **upgrade** command.
Step 16	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows the configuration steps for upgrading firmware for a Cisco Unified IP Phone 7960G or Cisco Unified IP Phone 7940G from SIP 5.3 to SIP 6.0, then from SIP 6.0 to SIP 7.4:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# load 7960 P0S3-06-0-00
Router(config-register-global)# upgrade
Router(config-register-global)# load 7960 P0S3-07-4-00
Router(config-register-global)# create profile
```

The following example shows the configuration steps for downgrading firmware for a Cisco Unified IP Phone 7960/40 from SIP 7.4 to SIP 6.0:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# load 7960 P0S3-06-0-00
Router(config-register-global)# upgrade
Router(config-register-global)# create profile
```

What to Do Next

- If Cisco Unified IP phones presently connected to Cisco Unified CME are using the SCCP protocol to receive and place calls and you now want some or all of these phones to use the SIP protocol, the phone firmware for each phone type must be upgraded from SCCP to the recommended SIP version before the phones can register. See the "Upgrading from SCCP to SIP" section on page 17.
- If Cisco Unified IP phones to be connected to Cisco Unified CME are brand new, out-of-the-box, the phone firmware preloaded at the factory must be upgraded to the recommended SIP version before your SIP phones can complete registration. See the "Upgrading from SCCP to SIP" section on page 17.
- If Cisco Unified IP phones presently connected to Cisco Unified CME are using the SIP protocol to receive and place calls and you now want some or all of these phones to use the SCCP protocol, the phone firmware for each phone type must be upgraded from SIP to the recommended SCCP version before the phones can register. See the "Upgrading from SIP to SCCP" section on page 20.

Upgrading from SCCP to SIP

To upgrade the Cisco phone firmware for a particular phone from SCCP to SIP, follow the steps in this task.

If Cisco Unified IP phones presently connected to Cisco Unified CME are using the SCCP protocol to receive and place calls and you now want some or all of these phones to use the SIP protocol, the phone firmware for each phone type must be upgraded from SCCP to the recommended SIP version before the phones can register. If Cisco Unified IP phones to be connected to Cisco Unified CME are brand new, out-of-the-box, the SCCP phone firmware preloaded at the factory must be upgraded to the recommended SIP version before your SIP phones can complete registration.

Note If codec values for the dial peers of a connection do not match, the call fails. The default codec for the POTS dial peer for an SCCP phone is G.711 and the default codec for a VoIP dial peer for a SIP phone is G.729. If neither the SCCP phone nor the SIP phone in Cisco Unified CME has been specifically configured to change the codec, calls between the two IP phones on the same router will produce a busy signal caused by the mismatched default codecs. To avoid codec mismatch, specify the codec for IP phones in Cisco Unified CME. For more information, see the"How to Configure Cisco Unified CME for Making and Receiving Basic Calls Using SIP Protocol" section on page 34.

Prerequisites

Cisco Unified IP Phone 7940Gs and Cisco Unified IP Phone 7960Gs—If these IP phones are already configured in Cisco Unified CME to use the SCCP protocol, the SCCP phone firmware on the phone must be version 5.x. If required, upgrade the SCCP phone firmware to 5.x before upgrading to SIP.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **no ephone** *ephone-tag*
4. **exit**

5. **no ephone-dn** *dn-tag*
6. **exit**
7. **voice register global**
8. **mode** *cme*
9. **load** *phone-type firmware-file*
10. **upgrade**
11. Repeat previous two steps.
12. **create profile**
13. **file text**
14. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`no ephone`** *`ephone-tag`* **Example:** `Router (config)# no ephone 23`	(Optional) Disables the ephone and removes the ephone configuration. • Required only if the Cisco Unified IP phone to be configured is already connected to Cisco Unified CME and is using SCCP protocol. • *ephone-tag*—Particular IP phone to which this configuration change will apply.
Step 4	**`exit`** **Example:** `Router(config-ephone)# exit`	(Optional) Exits from the current command mode to the next highest mode in the configuration mode hierarchy. • Required only if you performed the previous step.
Step 5	**`no ephone-dn`** *`dn-tag`*	(Optional) Disables the ephone-dn and removes the ephone-dn configuration. • Required only if this number or extension is not now nor will be associated to any SCCP phone line, intercom line, paging line, voice-mail port, or message-waiting indicator (MWI) connected to Cisco Unified CME. • *dn-tag*—Particular number or extension to which this configuration change will apply.

	Command or Action	Purpose
Step 6	**exit** **Example:** `Router(config-ephone-dn)# exit`	(Optional) Exits from the current command mode to the next highest mode in the configuration mode hierarchy. • Required only if you performed the previous step.
Step 7	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 8	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 9	**load** *phone-type firmware-file* **Example:** `Router(config-register-global)# load 7960-7940 P0S3-06-3-00`	Associates a phone type with a phone firmware file. • A separate **load** command is required for each IP phone type. • *phone-type*—Type **?** to display phone types or see the **load** command in the *Cisco Unified CME Command Reference*. • *firmware-file*—Filename to be associated with the specified Cisco Unified IP phone type. For Cisco ATA only, use the .sbin file extension. For all other Cisco Unified IP phone types, do not use the .sbin file extension.
Step 10	**upgrade** **Example:** `Router(config-register-global)# upgrade`	Generates a file with the universal application loader image for upgrading phone firmware and performs the TFTP server alias binding.
Step 11	Repeat previous two steps **Example:** `Router(config-register-global)# load 7960-7940 P0S3-07-4-00` `Router(config-register-global)# upgrade`	(Optional) Repeat for each version required in multistep upgrade sequences only.
Step 12	**create profile** **Example:** `Router(config-register-global;)# create profile`	Generates provisioning files required for SIP phones and writes the file to the location specified with the **tftp-path** command.

	Command or Action	Purpose
Step 13	**file text** **Example:** `Router(config-register-global)# file text`	(Optional) Generates ASCII text files for Cisco Unified IP Phones 7905 and 7905G, Cisco Unified IP Phone 7912 and Cisco Unified IP Phone 7912G, Cisco ATA-186, or Cisco ATA-188. By default, the system generates binary files to save disk space.
Step 14	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows the configuration steps for converting firmware on an Cisco Unified IP phone already connected in Cisco Unified CME and using the SCCP protocol, from SCCP 5.x to SIP 7.4:

```
Router(config)# telephony-service
Router(config-telephony)# no create cnf
CNF files deleted
Router(config-telephony)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# load 7960 P0S3-07-4-00
Router(config-register-global)# upgrade
Router(config-register-global)# create profile
```

What to Do Next

After you configure the **upgrade** command, refer to the following statements to determine which task to perform next.

- If the Cisco Unified IP phone to be upgraded is already connected in Cisco Unified CME and you removed the SCCP configuration file for the phone but have not configured the SIP phone in Cisco Unified CME, see the "Configuring SIP Phones in Cisco Unified CME" section on page 36.
- If all of the Cisco Unified IP phones in Cisco Unified CME are using the SIP protocol and you already configured the phone in Cisco Unified CME, see the "Resetting All SIP Phones After Upgrading Phone Firmware" section on page 24.
- If you have a combination of SIP and SCCP IP phones in Cisco Unified CME and you already configured the SIP phone in Cisco Unified CME, see the "Resetting an Individual SIP Phone After Upgrading Phone Firmware" section on page 26.

Upgrading from SIP to SCCP

To upgrade the Cisco phone firmware for a particular phone from SIP to SCCP, follow the steps in this task.

If Cisco Unified IP phones presently connected to Cisco Unified CME are using the SIP protocol to receive and place calls and you now want some or all of these phones to use the SCCP protocol, the phone firmware for each phone type must be upgraded from SIP to SCCP before the phones can register.

Note If codec values for the dial peers of a connection do not match, the call fails. The default codec for the POTS dial peer for an SCCP phone is G.711 and the default codec for a VoIP dial peer for a SIP phone is G.729. If neither the SCCP phone nor the SIP phone in Cisco Unified CME has been specifically configured to change the codec, calls between the two IP phones on the same router will produce a busy signal caused by the mismatched default codecs. To avoid codec mismatch, specify the codec for SIP and SCCP phones in Cisco Unified CME. For more information, see the "How to Configure Cisco Unified CME for Making and Receiving Basic Calls Using SIP Protocol" section on page 34.

Prerequisites

Cisco Unified IP Phone 7940Gs and Cisco Unified IP Phone 7960Gs—If these IP phones are already configured in Cisco Unified CME to use the SIP protocol, the SIP phone firmware must be version 7.x. See the "Upgrading or Downgrading Cisco Phone Firmware for SIP Phones Between Versions" section on page 13.

Removing a SIP Configuration Profile

To remove the SIP configuration profile before downloading the SCCP phone firmware to convert a phone from SIP to SCCP, perform the steps in this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **no voice register pool** *pool-tag*
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**no voice register pool** *pool-tag* **Example:** `Router(config)# no voice register pool 1`	Disables voice register pool and removes the voice pool configuration. • *pool-tag*—Unique sequence number for a particular SIP phone to which this configuration change will apply.
Step 4	**end** **Example:** `Router(config-register-pool)# exit`	Exits from the current command mode to the next highest mode in the configuration mode hierarchy.

Generating an SCCP XML Configuration File for Upgrading from SIP to SCCP

To create an ephone entry and generate a new SCCP XML configuration file for upgrading a particular Cisco Unified IP phone in Cisco Unified CME from SIP to SCCP, perform the steps in this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag*
4. **exit**
5. **tftp-server flash** *firmware-file*
6. **telephony service**
7. **load** *phone-type firmware-file*
8. **create cnf-files**
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**ephone-dn** *dn-tag* **Example:** `Router(config)# ephone dn 1`	Enters ephone-dn configuration mode, creates an ephone-dn, and optionally assigns it dual-line status. • *dn-tag*—Unique sequence number that identifies this ephone-dn during configuration tasks. The maximum number of ephone-dns in Cisco Unified CME is version and platform specific. Type **?** to display range.
Step 4	**exit** **Example:** `Router(config-ephone-dn)# exit`	Exits from the current command mode to the next highest mode in the configuration mode hierarchy.
Step 5	**tftp-server flash:** *file-name* **Example:** `Router(config)# tftp-server flash:P00307020300.loads` `Router(config)# tftp-server flash:P00307020300.sb2` `Router(config)# tftp-server flash:P00307020300.sbn` `Router(config)# tftp-server flash:P00307020300.bin`	Enables TFTP file sharing for new phone firmware files. • A separate **tftp-server flash** command is required for each firmware file to be downloaded to this phone.
Step 6	**telephony service** **Example:** `Router(config)# telephony service`	Enters telephone-service configuration mode.
Step 7	**load** *phone-type firmware-file* **Example:** `Router(config-telephony)# load 7960-7940 P00307020300`	Associates a phone type with a phone firmware file. • A separate **load** command is required for each IP phone type. • *phone-type*—Type **?** to display phone types. • *firmware-file*—Filename to be associated with the specified IP phone type. For ATA only, use the .sbin file extension. For all other Cisco Unified IP phone types, do not use the .sbin file extension.
Step 8	**create cnf-files** **Example:** `Router(config-telephony)# create cnf-files`	Builds XML configuration files required for SCCP phones.
Step 9	**end** **Example:** `Router(config-telephony)# exit`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows the configuration steps for upgrading firmware for a Cisco Unified IP Phone 7960G from SIP to SCCP. First the SIP firmware is upgraded to SIP 6.3 and from SIP 6.3 to SIP 7.4; then, the phone firmware is upgraded from SIP 7.4 to SCCP 7.2(3). The SIP configuration profile is deleted and a new ephone configuration profile is created for the Cisco Unified IP phone.

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# load 7960 P0S3-06-0-00
Router(config-register-global)# upgrade
Router(config-register-global)# load 7960 P0S3-07-4-00
Router(config-register-global)# exit
Router(config)# no voice register pool 1
Router(config-register-pool)# exit
Router(config)# voice register global
Router(config-register-global)# no upgrade
Router(config-register-global)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# exit
Router(config)# tftp-server flash:P00307020300.loads
Router(config)# tftp-server flash:P00307020300.sb2
Router(config)# tftp-server flash:P00307020300.sbn
Router(config)# tftp-server flash:P00307020300.bin
Router(config)# telephony service
Router(config-telephony)# load 7960-7940 P00307000100
Router(config-telephony)# create cnf-files
```

What to Do Next

After you configure the **upgrade** command:

- If the Cisco Unified IP phone to be upgraded is already connected in Cisco Unified CME and you removed the SIP configuration file for the phone and have not configured the SCCP phone in Cisco Unified CME, see the *Cisco Unified CallManager Express Administrator Guide* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.
- If you have a combination of SIP and SCCP IP phones in Cisco Unified CME and you already configured the SCCP phone in Cisco Unified CME, see the "Resetting SCCP Phones in Cisco Unified CME" section on page 28.

Resetting All SIP Phones After Upgrading Phone Firmware

To reset all SIP phones in Cisco Unified CME after you upgrade Cisco phone firmware from SCCP to SIP, and also return the **upgrade** command to its default, follow the steps in this task.

To reset an individual SIP phone after converting Cisco phone firmware, see the "Resetting an Individual SIP Phone After Upgrading Phone Firmware" section on page 26.

Prerequisites

A SIP configuration profile for each supported SIP phone in Cisco Unified CME is available for the phone to download from the location specified by the **tftp-path (voice register global)** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **reset**
6. **no upgrade**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**reset** **Example:** `Router(config-register-global)# reset`	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router and contacts the DHCP server and the TFTP server for updated information.

	Command or Action	Purpose
Step 6	**no upgrade** **Example:** `Router(config-register-global)# no upgrade`	Returns **upgrade** command to default condition.
Step 7	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Resetting an Individual SIP Phone After Upgrading Phone Firmware

To reset an individual SIP phone after you upgrade the Cisco phone firmware for that phone in Cisco Unified CME, and also return **upgrade** command to its default, follow the steps in this task.

Prerequisites

A SIP configuration profile for each supported SIP phone in Cisco Unified CME is available for the phone to download from the location specified by the **tftp-path (voice register global)** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **exit**
6. **voice register pool** *pool-tag*
7. **reset**
8. **exit**
9. **voice register global**
10. **no upgrade**
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**exit** **Example:** `Router(config-register-pool)# exit`	Exits from the current command mode to the next highest mode in the configuration mode hierarchy.
Step 6	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 7	**reset** **Example:** `Router(config-register-pool)# reset`	Performs a complete reboot of the single SIP phone specified with the **voice register pool** command and contacts the DHCP server and the TFTP server for updated information.
Step 8	**exit** **Example:** `Router(config-register-pool)# exit`	Exits from the current command mode to the next highest mode in the configuration mode hierarchy.

	Command or Action	Purpose
Step 9	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 10	**no upgrade** **Example:** `Router(config-register-global)# no upgrade`	Returns **upgrade** command to default condition.
Step 11	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Resetting SCCP Phones in Cisco Unified CME

This section contains the following tasks for performing a “hard” reboot of one or more SCCP phones in Cisco Unified CME:

- Resetting All SCCP Phones, page 28
- Resetting an Individual SCCP Phone, page 29

Resetting All SCCP Phones

To reset phones in all SCCP phones in Cisco Unified CME, follow the steps in this task.

Prerequisites

A SEP <MAC address> configuration file for each SCCP phone in Cisco Unified CME to be reset is available for the phone to download from the location specified by the **tftp-server flash** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony service**
4. **reset all**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony service** **Example:** `Router(config)# telephony service`	Enters telephony-service configuration mode.
Step 4	**reset all** **Example:** `Router(config-telephony)# reset all`	Performs a complete reboot of all SCCP phones associated with a Cisco Unified CME router and contacts the DHCP server and the TFTP server for updated information.
Step 5	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Resetting an Individual SCCP Phone

To reset an individual SCCP phone after you upgrade the Cisco phone firmware for that phone in Cisco Unified CME, follow the steps in this task.

Prerequisites

A SEP <MAC address> configuration file for each SCCP phone in Cisco Unified CME to be reset is available for the phone to download from the location specified by the **tftp-server flash** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *ephone-tag*
   or
   telephony-service
4. **reset**
   or
   reset *mac-address*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone** *ephone-tag* or **telephony-service** **Example:** `Router(config)# ephone 1` or `Router(config)# telephony-service`	Enters ephone configuration mode. • *ephone-tag*—Identifies IP phone to be configured. or Enters telephony-service configuration mode.
Step 4	**reset** or **reset** *mac-address* **Example:** `Router(config-ephone)# reset` or `Router(config-telephony-service)# reset CFBA.321B.96FA`	Performs a complete reboot of the IP phone specified by the **ephone** command and contacts the DHCP server and the TFTP server for updated information. or Performs a complete reboot of the IP phone with the specified MAC address and contacts the DHCP server and the TFTP server for updated information.
Step 5	**end** **Example:** `Router(config-ephone)# end` or `Router(config-telephony-service)# end`	Exits configuration mode and enters privileged EXEC mode.

Verifying the Phone Firmware Version on an IP Phone

Use the **show ephone phone-load** command to verify which phone firmware is installed on a particular ephone. The DeviceName includes the MAC address for the IP phone.

```
Router# show ephone phone-load

DeviceName        CurrentPhoneload       PreviousPhoneload      LastReset
=====================================================================
SEP000A8A2C8C6E   7.3(3.02)                                     Initialized
```

Troubleshooting Tips

Use the **debug tftp event** command to troubleshoot an attempt to upgrade or convert Cisco phone firmware files for SIP phones. The following sample from the **debug tftp event** command shows how the Cisco phone firmware for a Cisco Unified IP Phone 7940G is upgraded from SCCP 5.X to SIP 6.3. The configuration profiles are downloaded when a phone is rebooted or reset.

```
Router# debug tftp event
…
Router(config)#telephony-service
Router(config-telephony)#no create cnf
CNF files deleted
Router(config-telephony)#voice register global
Router(config-register-global)#load 7960 P0S3-06-3-00
Router(config-register-global)#upgrade
Router(config-register-global)#create profile
Router(config-register-global)#
*May 6 17:37:03.737: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000ED7DF7932 IP:1.5.49.84
Socket:4
DeviceType:Phone has unregistered normally.
*May 6 17:37:35.949: TFTP: Looking for OS79XX.TXT
*May 6 17:37:36.413: TFTP: Opened system:/cme/sipphone/OS79XX.TXT, fd 4, size 13 for
process 81
*May 6 17:37:36.413: TFTP: Finished system:/cme/sipphone/OS79XX.TXT, time 00:00:00 for
process 81
*May 6 17:37:40.533: TFTP: Looking for P0S3-06-3-00.sbn
*May 6 17:37:40.541: TFTP: Opened flash:P0S3-06-3-00.sbn, fd 4, size 487198 for process 81
*May 6 17:37:48.225: TFTP: Finished flash:P0S3-06-3-00.sbn, time 00:00:07 for process 81
*May 6 17:40:26.925: TFTP: Looking for OS79XX.TXT
*May 6 17:40:26.925: TFTP: Opened system:/cme/sipphone/OS79XX.TXT, fd 4, size 13 for
process 81
*May 6 17:40:26.925: TFTP: Finished system:/cme/sipphone/OS79XX.TXT, time 00:00:00 for
process 81
*May 6 17:40:26.929: TFTP: Looking for SIPDefault.cnf
*May 6 17:40:26.929: TFTP: Opened system:/cme/sipphone/SIPDefault.cnf, fd 4, size 1558 for
process 81
*May 6 17:40:26.937: TFTP: Finished system:/cme/sipphone/SIPDefault.cnf, time 00:00:00 for
process 81
*May 6 17:40:27.053: TFTP: Looking for SIP000ED7DF7932.cnf
*May 6 17:40:27.053: TFTP: Opened system:/cme/sipphone/SIP000ED7DF7932.cnf, fd 4, size 789
for process 81
*May 6 17:40:27.057: TFTP: Finished system:/cme/sipphone/SIP000ED7DF7932.cnf, time
00:00:00 for process 81
```

The following sample from the **debug tftp event** command shows how the Cisco phone firmware for a Cisco Unified IP Phone 7940G is upgraded from SIP 6.3 to SIP 7.0 after the phone is rebooted or reset:

```
Router# debug tftp event
…
Router(config-register-global)#load 7960 P003-07-4-00
Router(config-register-global)#upgrade
Router(config-register-global)#load 7960 P0S3-07-4-00
Router(config-register-global)#create profile
Router(config-register-global)#end
Router-2012#
*May 6 17:42:35.581: TFTP: Looking for OS79XX.TXT
*May 6 17:42:35.585: TFTP: Opened system:/cme/sipphone/OS79XX.TXT, fd 5, size 13 for
process 81
*May 6 17:42:35.585: TFTP: Finished system:/cme/sipphone/OS79XX.TXT, time 00:00:00 for
process 81
*May 6 17:42:35.969: TFTP: Looking for P003-07-4-00.sbn
*May 6 17:42:35.977: TFTP: Opened slot0:P003-07-4-00.sbn, fd 5, size 129876 for process 81
```

```
*May 6 17:42:37.937: TFTP: Finished slot0:P003-07-4-00.sbn, time 00:00:01 for process 81
*May 6 17:44:31.037: TFTP: Looking for CTLSEP000ED7DF7932.tlv
*May 6 17:44:31.057: TFTP: Looking for SEP000ED7DF7932.cnf.xml
*May 6 17:44:31.089: TFTP: Looking for SIP000ED7DF7932.cnf
*May 6 17:44:31.089: TFTP: Opened system:/cme/sipphone/SIP000ED7DF7932.cnf, fd 5, size 789
for process 81
*May 6 17:44:31.089: TFTP: Finished system:/cme/sipphone/SIP000ED7DF7932.cnf, time
00:00:00 for process 81
*May 6 17:44:31.125: TFTP: Looking for P0S3-07-4-00.loads
*May 6 17:44:31.133: TFTP: Opened slot0:P0S3-07-4-00.loads, fd 5, size 461 for process 81
*May 6 17:44:31.141: TFTP: Finished slot0:P0S3-07-4-00.loads, time 00:00:00 for process 81
*May 6 17:44:31.673: TFTP: Looking for P0S3-07-4-00.sb2
*May 6 17:44:31.681: TFTP: Opened slot0:P0S3-07-4-00.sb2, fd 5, size 592626 for process 81
*May 6 17:44:33.989: TFTP: Finished slot0:P0S3-07-4-00.sb2, time 00:00:02 for process 81
```

Configuring Cisco Unified CME for Making Basic Calls Using SIP Phones

Revised: June 19, 2006

This chapter describes how to configure Cisco CME 3.4 and later to support making basic calls using SIP phones connected directly in Cisco Unified CallManager Express (Cisco Unified CME).

Finding Feature Information

Your Cisco IOS software release may not support all of the features documented in this guide. To reach links to specific feature documentation in this guide and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Unified CME for SIP Phones" section in this guide.

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at
http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Finding Support Information for Cisco Unified CME

For information about Cisco IOS software and Cisco Unified CME compatibility, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

For information about Cisco Unified CME specifications, including number of supported phones, see the appropriate *Cisco Unified CME Firmware, Platforms, Memory, and Voice Products* document at
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Contents

Prerequisites for Configuring Cisco Unified CME for Making Basic Calls Using SIP Phones

- If applicable, PSTN lines are configured and operational.
- If applicable, the WAN links are configured and operational.
- SCCP and SIP phone firmware for Cisco Unified IP phones to be connected to Cisco Unified CME, including all releases required during an upgrade sequence, is installed in the flash memory of the TFTP server from which the SIP phones will download their configuration profiles.
- Cisco phone firmware files are upgraded as required. See the "Upgrading, Downgrading, and Converting Cisco Phone Firmware Files" section on page 11.

Restrictions for Configuring Cisco Unified CME for Making Basic Calls Using SIP Phones

All restrictions described in the *Cisco Unified CallManager Express System Administrator Guide* apply to SIP phone support; see http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

How to Configure Cisco Unified CME for Making and Receiving Basic Calls Using SIP Protocol

Note This guide includes information for configuring Cisco Unified CME only and does not include procedures for configuring SIP in Cisco Unified SRST. To find information about configuring SIP in Cisco Unified SRST, see the *Cisco Unified SRST System Administrator Guide* at: http://www.cisco.com/en/US/products/sw/voicesw/ps2169/products_feature_guide09186a008018912f.html.

This section includes the following tasks:

- Resetting SIP Phones in Cisco Unified CME, page 60
- Verifying the Configuration, page 62

Enabling SIP-to-SIP Calls in Your VoIP Network

To enable incoming and outgoing calls between SIP phones in Cisco Unified CME and to enable SIP registrar functionality on your Cisco Unified CME router, follow the steps in this section.
Cisco Unified CME supports Media Flow-through mode only; enabling SIP-to-SIP calls is required before you can successfully make SIP-to-SIP calls.

Tip Cisco Unified CME does not maintain a persistent database of registration entries across reloads. If the WAN is down, and you reboot your Cisco Unified CME router, when the router reloads it will have no database of SIP phone registrations. The SIP phones will have to register again, which could take several minutes, as SIP phones do not use a keepalive functionality. To shorten the time before the phones reregister, use the [**expires** [**max** *sec*] keyword and argument combination for the **registrar server** command to adjust the registration expiry. The default expiry is 3600 seconds; we recommend an expiry of 600.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from*-**type to** *to*-**type**
5. **sip**
6. **registrar server** [**expires** [**max** *sec*][**min** *sec*]
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode and specifies Voice over IP (VoIP) encapsulation.
Step 4	**allow-connections** *from*-**type to** *to*-**type** **Example:** `Router(config-voi-srv)# allow-connections SIP to SIP`	Enables calls between specific types of endpoints in a VoIP network. • *from*-**type**—Valid endpoint type is SIP. • *to*-**type**—Valid endpoint type is SIP.
Step 5	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 6	**registrar server** [**expires** [**max** *sec*][**min** *sec*]] **Example:** `Router(config-voi-sip)# registrar server expires max 600 min 60`	Enables SIP registrar functionality in Cisco Unified CME. • **expires**—(Optional) Sets the active time for an incoming registration. • **max** *sec*—(Optional) Maximum time for a registration to expire, in seconds. Range is from 600 to 86400. Default is 3600. Recommended value is 600. • **min** *sec*—(Optional) Minimum time for a registration to expire, in seconds. Range is from 60 to 3600. Default is 60.
Step 7	**end** **Example:** `Router(config-voi-sip)# end`	Exits configuration mode and enters privileged EXEC mode.

Configuring SIP Phones in Cisco Unified CME

The tasks in this section are required to create or modify configuration parameters so that SIP phones can automatically find the defaults to configure themselves when they come online or are rebooted.

Prerequisites

- VoIP network is enabled for SIP-to-SIP calling.
- SIP registrar functionality is enabled in Cisco Unified CME. See the "Enabling SIP-to-SIP Calls in Your VoIP Network" section on page 35.

Specifying Required System-Wide Parameters

To set system-wide (global) parameters to be applied to all individual directory numbers and all numbers on SIP phones in Cisco Unified CME.

Note If your Cisco Unified CME system supports SCCP and also SIP phones, do *not* connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **source-address** *ip-address*
6. **load** *phone-type firmware-file*
7. **max-dn** *max-directory-numbers*
8. **max-pool** *max-phones*
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	`enable` **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	`configure terminal` **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**source-address** *ip-address* [**port** *port*] **Example:** `Router(config-register-global)# source-address 10.6.21.4`	Enables a router to receive messages from SIP phones through the specified IP address and port. • Preexisting router *ip-address*, typically an address of one of the Ethernet ports of the Cisco Unified CME router. • **port**—(Optional) TCP/IP *port* number. Range is 2000 to 9999. Default is 2000.
Step 6	**load** *phone-type firmware-file* **Example:** `Router(config-register-global)# load 7960-7940 P0S3-07-3-00`	Associates a phone type with a phone firmware file. • A separate **load** command is required for each IP phone type. • *firmware-file*—Filename to be associated with the specified Cisco Unified IP phone type. For Cisco ATA only, use the .sbin file extension. For all other Cisco Unified IP phone types, do not use the .sbin file extension.
Step 7	**max-dn** *max-directory-numbers* **Example:** `Router(config-register-global)# max-dn 20`	(Optional) Limits the maximum number of directory numbers for SIP phones to be supported by the Cisco Unified CME router. • Maximum number of directory numbers supported by a router. The maximum number is version and platform dependent. Type **?** to display value. • Range is 1 to 150 or maximum number supported by the Cisco Unified router. Default is 150.
Step 8	**max-pool** *max-phones* **Example:** `Router(config-register-global)# max-pool 10`	(Optional) Limits the maximum number of SIP phones to be supported the by the Cisco Unified CME router. • Maximum number of phones supported by a router. The maximum number is version and platform dependent. Type **?** to display value. • Default is 0.
Step 9	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Setting Date and Time Parameters for SIP Phones

To specify the format of the date and time that appears on all SIP phones in Cisco Unified CME, follow the steps in this section.

Note If your Cisco Unified CME system supports SCCP and also SIP phones, do *not* connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **date-format** [**d/m/d** | **m/d/y** | ***y-d-m*** | **y/d/m** | **y/m/d** | ***yy-m-d***]
5. **time-format** {**12** | **24**}
6. **timezone** *number*
7. **dst auto-adjust**
8. **dst** {**start** | **stop**} *month* [**day** *day-of-month* | **week** *week-number* | **day** *day-of-week*] **time** *hour:minutes*
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**date-format** [d/m/d | m/d/y | *y-d-m* | y/d/m | y/m/d | *yy-m-d*] **Example:** `Router(config-register-global)# date-format dd-mm-yy`	(Optional) Selects the date display format on SIP phones in Cisco Unified CME. • Default is **m-d-y**.

	Command or Action	Purpose
Step 5	**time-format** {**12** | **24**} **Example:** `Router(config-register-global)# time-format 24`	(Optional) Selects the time display format on SIP phones in Cisco Unified CME. • Default is **12**.
Step 6	**timezone** *number* **Example:** `Router(config-register-global)# timezone 8`	Selects the time zone used for SIP phones in Cisco Unified CME. • Default is **5**, Pacific Standard/Daylight Time. • Type **?** to display a list of time zones.
Step 7	**dst auto-adjust** **Example:** `Router(config-register-global)# dst auto-adjust`	(Optional) Enables automatic adjustment of daylight savings time on SIP phones in Cisco Unified CME. • To modify start and stop times for daylight savings time, use the **dst** command.
Step 8	**dst** {**start** | **stop**} *month* [**day** day-of-month | **week** week-number | **day** day-of-week] **time** *hour:minutes* **Example:** `Router(config-register-global)# dst start jan day 1 time 00:00` `Router(config-register-global)# dst stop mar day 31 time 23:59`	(Optional) Sets the time period for daylight savings time on SIP phones in Cisco Unified CME. • Required if automatic adjustment of daylight savings time is enabled by using the **dst auto-adjust** command. • Default is Start: First week of April, Sunday, 2:00 a.m and Stop: Last week of October, Sunday 2:00 a.m. • *month*—Valid abbreviations are: **jan**, **feb**, **mar**, **apr**, **may, jun**, **jul**, **aug**, **sep, oct, nov, dec.** • **day** *day-of-month*—Range is 1 to 31. • **week** *week-number*—Range is 1 to 4, or 8. 8 represents the last week of the month. • **day** *day-of-week*—Valid abbreviations are: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**, **sun.** • **time** *hour:minutes* —Beginning and ending times for daylight savings time in an HH:MM format using a 24-hour clock. The stop time must be greater than the start-time. The value 24:00 is not valid. If you enter 00:00 as a stop time, it is changed to 23:59. If you enter 00:00 for both start time and stop time, daylight savings time is enabled for the entire 24-hour period on the specified day.
Step 9	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Specifying Optional System-Wide Parameters

To set optional system-wide (global) parameters to be applied to all individual directory numbers and all numbers on SIP phones in Cisco Unified CME.

Note If your Cisco Unified CME system supports SCCP and also SIP phones, do *not* connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **logo** *url*
5. **mwi reg-e164**
6. **mwi stutter**
7. **phone-redirect-limit** *number*
8. **application** *application-name*
9. **authenticate** [**all**][**realm** *string*]
10. **url** {**directory** | **service**} *url*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	`enable` **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	`configure terminal` **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	`voice register global` **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	`logo` *`url`* **Example:** `Router(config-register-global)# logo http://mycompany.com/files/logo.xml`	(Optional) Specifies a file to display on SIP phones in Cisco Unified CME. • url—URL as defined in RFC 2396. The file that is displayed must be encoded in eXtensible Markup Language (XML) by using the Cisco XML document type definition (DTD)
Step 5	`mwi reg-e164` **Example:** `Router(config-register-global)# mwi reg-e164`	(Optional) Registers full E.164 number to the MWI server in Cisco Unified CME and enables MWI.

	Command or Action	Purpose
Step 6	**`mwi stutter`** **Example:** `Router(config-register-global)# mwi stutter`	(Optional) Enables Cisco Unified CME router at the central site to relay MWI notification to remote SIP phones.
Step 7	**`phone-redirect-limit`** *`number`* **Example:** `Router(config-register-global)# phone-redirect-limit 8`	(Optional) Changes the default number of 3XX responses a SIP phone that originates a call can handle for a single call. • *number*—Maximum number of 3XX responses accepted for a single call. Range is 5 to 20. Default is 5.
Step 8	**`application`** *`application-name`* **Example:** `Router(config-register-global)# application sipapp2`	(Optional) Changes the default application for all dial peers associated with the SIP phones in Cisco Unified CME to the specified application. **Note** The **application** command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode. • *application-name*—Interactive voice response (IVR) application name. The applied application (or Tcl IVR script) must support call redirection.
Step 9	**`authenticate`** `[`**`all`**`][`**`realm`** *`string`*`]` **Example:** `Router(config-register-global)# authenticate all realm company.com`	(Optional) Enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods. Required only if you want to enable authentication. • **all**—(Optional) Causes all registrations requests to be challenged. • **realm** *string*—(Optional) Keyword and argument combination for challenge and response as specified in RFC 2617.

	Command or Action	Purpose
Step 10	**url** {**directory** | **service**} *url* **Example:** `Router(config-register-global)# url directory http://10.0.0.11/localdirectory` `Router(config-register-global)# url service http://10.0.0.4/CCMUser/123456/urltest.html`	(Optional) Associates a URL with the programmable Directories and Services feature buttons on Cisco Unified IP Phone 7960s and 7960Gs and Cisco Unified IP Phone 7940s and 7940Gs in Cisco Unified CME. Operation of these services is determined by the Cisco Unified IP phone capabilities and the content of the specified URL. **Note** Provisioning the directory URL to select an external directory resource disables the Cisco Unified CME local directory service. • **directory**—Uses the information at the specified URL for the Directories button display. Disable the the local directory by specifying "none" instead of a URL with the **directory** keyword or **service**—Uses the information at the specified URL for the Services button display. • *url*—URL as defined in RFC 2396.
Step 11	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Creating Directory Numbers for SIP phones

To create a directory number (DN) in Cisco Unified CME for a SIP phone, intercom line, voice port, or a message-waiting indicator (MWI), follow the steps in this section for each directory number to be created.

Prerequisites

The maximum number of directory numbers supported by a router is version and platform dependent. To configure a maximum number of directory numbers to some number other than the default of 150, use the **max-dn (voice register global)** command before performing this task. See the "Specifying Required System-Wide Parameters" section on page 37.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number** *number*
5. **preference** *preference-order*
6. **name** *name*
7. **label** *string*
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register dn** *dn-tag* **Example:** `Router(config-register-global)# voice register dn 17`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or a message-waiting indicator (MWI). • d*n-tag*—Identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn (voice register global)** command.
Step 4	**number** *number* **Example:** `Router(config-register-dn)# number 7001`	Defines a valid number for a directory number. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string may contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.
Step 5	**preference** *preference-order* **Example:** `Router(config-register-dn)# preference 2`	Indicates the preference order for matching dial peers in a virtual dial-peer group. • *preference-order*—Order in which dial peer is selected when multiple dial peers are matched on the same destination pattern. Range is 0 to 10. Default is 0, which is the highest preference.
Step 6	**name** *name* **Example:** `Router(config-register-dn)# name Smith, John` or **Example:** `Router(config-register-dn)# name John Smith`	(Optional) Associates a name with a directory number in Cisco Unified CME and provides caller ID for calls originating from a SIP phone. • Name must follow the order specified in the **directory (telephony-service)** command, either **first-name-last** or **last-name-first**.

	Command or Action	Purpose
Step 7	`label` *string* **Example:** `Router(config-register-dn)# label user01`	(Optional) Creates a text identifier, instead of a phone-number display, for a directory number that appears on a SIP phone console. • One label is permitted per extension. • Directory number must already have a number assigned by using the **number (voice register dn)** command.
Step 8	`end` **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

What to Do Next

- If you are adding one or more new SIP phones to Cisco Unified CME, you are ready to specify phone-specific parameters for the configuration profile. See the "Specifying Phone-Specific Parameters for SIP Phones in Cisco Unified CME" section on page 45.
- If you are modifying parameters for SIP phones already in Cisco Unified CME and you want to change phone-specific parameters for an individual phone, see the "Specifying Phone-Specific Parameters for SIP Phones in Cisco Unified CME" section on page 45.
- If you are finished modifying parameters for SIP phones already in Cisco Unified CME, generate a new configuration profile to propagate the modifications. See the "Generating Configuration Profiles for SIP Phones" section on page 51.

Specifying Phone-Specific Parameters for SIP Phones in Cisco Unified CME

To create and modify phone-specific parameters in SIP configuration profiles, follow the steps in this section for each SIP phone to be connected in Cisco Unified CME.

Note If your Cisco Unified CME system supports SCCP and also SIP phones, do *not* connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **exit**
6. **voice register pool** *pool-tag*
7. **id mac** *address*
8. **type** *phone-type*
9. **number** *tag* {*number-pattern* [**preference** *value* [**huntstop**]] | **dn** *dn-tag*}

10. **codec** *codec-type* [*bytes*]
    or
    voice-class codec *tag*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice register global`** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**`mode`** *`cme`* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**`exit`** **Example:** `Router(config-register-global)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 6	**`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones in Cisco Unified CME. • *pool-tag*—Identifies a particular SIP phone during configuration tasks. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 7	**`id`** {**`network`** *`address`* **`mask`** *`mask`* | **`ip`** *`address`* **`mask`** *`mask`* | **`mac`** *`address`*} **Example:** `Router(config-register-pool)# id mac 0009.A3D4.1234`	Explicitly identifies a locally available individual SIP phone to support a degree of authentication. • **network** *address* **mask** *mask*—Keyword/argument combination used to accept SIP Register messages for the indicated phone numbers from any IP phone within the specified IP subnet. • **ip** *address* **mask** *mask*—Unique IP address of an individual IP phone. • **mac** *address*—Unique MAC address of an individual IP phone.

	Command or Action	Purpose
Step 8	**type** *phone-type* **Example:** `Router(config-register-pool)# type 7960-7940`	Defines a phone type for the SIP phone being configured. • *phone-type*—Type **?** to display a list of phone types.
Step 9	**number** *tag* {*number-pattern* [**preference** *value* [**huntstop**]] | **dn** *dn-tag*} **Example:** `Router(config-register-pool)# number 1 4085557...` or `Router(config-register-pool)# number 1 dn 17`	Associates a phone number or a directory number with the SIP phone being configured. • *tag*—Identifies the number being configured in a list of numbers for this pool. Range is 1 to 10. • *number-pattern*—Identifies phone numbers (including wild cards and patterns) that are permitted by the registrar to handle the Register message from this SIP phone. • **preference** *value*—(Optional) Defines the preference order of the number. Range is 0 to 10 with 0 being the highest preference. There is no default. • **huntstop—**(Optional) Stops hunting if the dial peer is busy. • **dn** *dn-tag*—Identifies the directory number for this SIP phone as defined by the **voice register dn** command. Range is 1 to 150.

	Command or Action	Purpose
Step 10	`codec` *`codec-type`* `[`*`bytes`*`]` or **`voice-class codec`** *`tag`* **Example:** `Router(config-register-pool)# codec g711alaw` or `Router(config-register-pool)# voice-class codec 1`	Specifies the codec for the dial peer dynamically created when the SIP phone registers. If codec values for the dial peers of an internal connection do not match, the call fails. **Note** This command overrides any previously configured codec selection set with the **voice-class codec** command. **Tip** If G.729 is the desired codec for Cisco ATA-186 and Cisco ATA-188, then only one port of the Cisco ATA device should be configured in Cisco Unified CME. If a call is placed to the 2nd port of the Cisco ATA device, it will be disconnected gracefully. If a you want to use both Cisco ATA ports simultaneously, then configure G.711 in Cisco Unified CME. • *codec-type*—Valid entries are as follows: – **g711alaw**—G.711a-law 64,000 bps. – **g711ulaw**—G.711u-law 64,000 bps. – **g729r8**—G.729 8000 bps. • *bytes*—(Optional) Specifies the number of bytes in the voice payload of each frame. or Assigns a previously configured codec selection preference list to change the automatically selected default codec for the dial peer dynamically created when the SIP phone registers. If codec values for the dial peers of an internal connection do not match, the call fails. **Note** Voice-class with codec list can be configured, but more than one list member can not be supported for B2BUA call. *tag*—Unique number assigned to the voice class. Range is from 1 to10000. The tag number maps to the tag number created by using the **voice class codec (dial peer)** command.
Step 11	**`end`** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Specifying Optional Phone-Specific Parameters for SIP Phones in Cisco Unified CME

To create and modify optional phone-specific parameters in the SIP configuration profiles, follow the steps in this section for each SIP phone to be connected in Cisco Unified CME.

Note If your Cisco Unified CME system supports SCCP and also SIP phones, do *not* connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **preference** *preference-order*
5. **description** *string*
6. **username** *name* **password** *string*
7. **call-waiting**
8. **cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* [**-** *ending-number]* | **default**}
9. **dtmf-relay[rtp-nte]**
10. **application** *application-name*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	`enable` **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	`configure terminal` **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	`voice register pool` *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones in Cisco Unified CME. • *pool-tag*—Identifies a particular SIP phone during configuration tasks. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 4	`preference` *preference-order* **Example:** `Router(config-register-pool)# preference 4`	(Optional) Creates preference order for the VoIP dial peers created for a number associated with a voice pool to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern and establish a hunt strategy for incoming calls. • Preference order for the number associated with a pool. Range is 0 to 10. Default is 0, which is the highest preference.

	Command or Action	Purpose
Step 5	`description` *string* **Example:** `Router(config-register-pool)# description 408-555-0100`	(Optional) Defines a customized description that appears in the header bar of Cisco Unified IP Phone 7940 and 7940G, and Cisco Unified IP Phone 7960 and 7960G. • *string*—Up to 40 alphanumeric characters. Truncated to 14 characters in the display. **Note** If string contains spaces, enclose the string in quotation marks.
Step 6	**username** user*name* **password** *string* **Example:** `Router(config-register-pool)# username smith password 123zyx`	(Optional) Required only if authentication is enabled with the **authenticate** command. Creates an authentication credential for SIP phone registration only. **Note** This command is not for SIP proxy registration. The password will not be encrypted. All lines in a phone will share the same credential. • *username*—Identifies a local Cisco Unified IP phone user. Default is Admin. • **password**—Enables a password for the local Cisco Unified IP phone user. • Enter *password* string.
Step 7	**call-waiting** **Example:** `Router(config-register-pool)# call-waiting`	(Optional) Configures Call Waiting feature on the SIP phone being configured. **Note** This step is included to illustrate how to enable the command if it was previously disabled. • Default is enabled.
Step 8	**cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* [**-** *ending-number*] | **default**} **Example:** `Router(config-register-pool)# cor incoming call91 1 91011`	(Optional) Configures a class of restriction (COR) for the dynamically created VoIP dial peers associated with directory numbers and specifies which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list. • **incoming**—COR list to be used by incoming dial peers. • **outgoing**—COR list to be used by outgoing dial peers. • *cor-list-name*—Name of COR list. • *cor-list-number*—Number of COR list. • s*tarting-number*—Start of a directory number range, if an ending number is included. Can also be a standalone number. • **-***ending-number*—(Optional) End of a directory number range. The hyphen (-) is required before the ending number to indicate that a full range is to be configured. • **default**—Instructs the COR list to assume behavior according to a predefined default COR list.

	Command or Action	Purpose
Step 9	**dtmf-relay** [**cisco-rtp**] [**rtp-nte**] [**sip-notify**] **Example:** `Router(config-register-pool)# dtmf-relay rtp-nte`	(Optional) Specifies a list of DTMF relay methods that can be used by a SIP phone to relay DTMF tones. **Note** SIP phones in Cisco Unified CME support RFC2833 (RTP-NTE) only. • **rtp-nte**—Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Named Telephone Event (NTE) payload.
Step 10	**application** *application-name* **Example:** `Router(config-register-pool)# application sipapp2`	(Optional) Changes the default application for all dial peers associated with the voice pool in Cisco Unified CME to the specified application. • *application-name*—Interactive voice response (IVR) application name. The applied application (or Tcl IVR script) must support call redirection.
Step 11	**end** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Generating Configuration Profiles for SIP Phones

To generate the configuration profile files that are required by the SIP phones in Cisco Unified CME and write them to the location specified by the **tftp-path (voice register global)** command, follow the steps in this section.

Any time you create or modify parameters under the voice register dn or voice register pool configuration modes, use the **create profile** command to generate a new configuration profile and properly propagate the parameters.

Caution If your Cisco Unified CME system supports SCCP and also SIP phones, do *not* connect your SIP phones to the network until after you have verified the phone configuration profiles.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **tftp path** *path*
6. **file text** (Optional)
7. **create profile**
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**tftp-path** *path* **Example:** `Router(config-register-global)# tftp-path http://mycompany.com/files`	Defines the location from which the SIP phones will download configuration profile files. • *path*—Valid entries are as follows: – **system:/cme/sipphone** (default)—System memory in the Cisco Unified CME router – **flash:**—Router flash memory – **slot:** – **http://**—External TFTP server
Step 6	**file text** **Example:** `Router(config-register-global)# file text`	(Optional) Generates ASCII text files of the configuration profiles generated for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186, or Cisco ATA-188.
Step 7	**create profile** **Example:** `Router(config-register-global;)# create profile`	Generates configuration profile files required for SIP phones and writes the files to the location specified with the **tftp-path** command.
Step 8	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Verifying the Configuration Profiles

Before connecting or resetting SIP phones to Cisco Unified CME, verify the configuration profiles generated by completing all previous tasks in this section. If the configuration is correct, the SIP phone to be connected to Cisco Unified CME will be able to register and minimally, have an assigned phone number.

Step 1 **voice register tftp-bin**

Use the **voice register tftp-bind** command to display a list of configuration profiles that are accessible to SIP phones using TFTP. The file name includes the MAC address for each SIP phone, such as SIP<mac-address>.cnf. Verify that a configuration profile is available for each SIP phone in Cisco Unified CME.

The following is sample output from this command:

```
Router(config)# show voice register tftp-bind
 tftp-server SIPDefault.cnf url system:/cme/sipphone/SIPDefault.cnf
 tftp-server syncinfo.xml url system:/cme/sipphone/syncinfo.xml
 tftp-server SIP0009B7F7532E.cnf url system:/cme/sipphone/SIP0009B7F7532E.cnf
 tftp-server SIP000ED7DF7932.cnf url system:/cme/sipphone/SIP000ED7DF7932.cnf
 tftp-server SIP0012D9EDE0AA.cnf url system:/cme/sipphone/SIP0012D9EDE0AA.cnf
 tftp-server gk123456789012 url system:/cme/sipphone/gk123456789012
 tftp-server gk123456789012.txt url system:/cme/sipphone/gk123456789012.txt
```

Step 2 **show voice register profile**

Use the **show voice register profile** command to display the contents of the ASCII format configuration profile for a particular voice register pool.

Note To generate ASCII text files of the configuration profiles for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, and Cisco ATA-188s, use the **file text** command.

The following is sample output from this command displaying information in the configuration profile for voice register pool 4.

```
Router# show voice register profile text 4
 Pool Tag: 4
# txt
 AutoLookUp:0
 DirectoriesUrl:0
…
 CallWaiting:1
 CallForwardNumber:0
 Conference:1
 AttendedTransfer:1
 BlindTransfer:1
…
 SIPRegOn:1
 UseTftp:1
 UseLoginID:0
 UIPassword:0
 NTPIP:0.0.0.0
 UID:2468
…
```

Step 3 **more system**

Use the **more system** command to display the contents of the configuration profile for a particular Cisco Unified IP Phone 7940, Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7960, or Cisco Unified IP Phone 7960G.

The following is sample output from this command displaying information in two SIP configuration profile files. The SIPDefault.cnf configuration profile is a shared file and SIP<MAC address>.cnf is the SIP configuration profile for the SIP phone with the designated MAC address.

```
Router# more system:/cme/sipphone/SIPDefault.cnf
image_version: "P0S3-07-4-00";
proxy1_address: "10.1.18.100";
proxy2_address: "";
proxy3_address: "";
proxy4_address: "";
proxy5_address: "";
proxy6_address: "";
proxy1_port: "5060";
proxy2_port: "";
proxy3_port: "";
proxy4_port: "";
proxy5_port: "";
proxy6_port: "";
proxy_register: "1";
time_zone: "EST";
dst_auto_adjust: "1";
dst_start_month: "April";
dst_start_day: "";
dst_start_day_of_week: "Sun";
dst_start_week_of_month: "1";
dst_start_time: "02:00";
dst_stop_month: "October";
dst_stop_day: "";
dst_stop_day_of_week: "Sun";
dst_stop_week_of_month: "8";
dst_stop_time: "02:00";
date_format: "M/D/Y";
time_format_24hr: "0";
local_cfwd_enable: "1";
directory_url: "";
messages_uri: "2000";
services_url: "";
logo_url: "";
stutter_msg_waiting: "0";
sync: "0000200155330856";
telnet_level: "1";
autocomplete: "1";
call_stats: "0";
Domain_Name: "";
dtmf_avt_payload: "101";
dtmf_db_level: "3";
dtmf_inband: "1";
dtmf_outofband: "avt";
dyn_dns_addr_1: "";
dyn_dns_addr_2: "";
dyn_tftp_addr: "";
end_media_port: "32766";
http_proxy_addr: "";
http_proxy_port: "80";
nat_address: "";
nat_enable: "0";
nat_received_processing: "0";
network_media_type: "Auto";
network_port2_type: "Hub/Switch";
```

```
outbound_proxy: "";
outbound_proxy_port: "5060";
proxy_backup: "";
proxy_backup_port: "5060";
proxy_emergency: "";
proxy_emergency_port: "5060";
remote_party_id: "0";
sip_invite_retx: "6";
sip_retx: "10";
sntp_mode: "directedbroadcast";
sntp_server: "0.0.0.0";
start_media_port: "16384";
tftp_cfg_dir: "";
timer_invite_expires: "180";
timer_register_delta: "5";
timer_register_expires: "3600";
timer_t1: "500";
timer_t2: "4000";
tos_media: "5";
voip_control_port: "5060";

Router# more system:/cme/sipphone/SIP000CCE62BCED.cnf
image_version: "P0S3-07-4-00";
user_info: "phone";
line1_name: "1051";
line1_displayname: "";
line1_shortname: "";
line1_authname: "1051";
line1_password: "ww";
line2_name: "";
line2_displayname: "";
line2_shortname: "";
line2_authname: "";
line2_password: "";
auto_answer: "0";
speed_line1: "";
speed_label1: "";
speed_line2: "";
speed_label2: "";
speed_line3: "";
speed_label3: "";
speed_line4: "";
speed_label4: "";
speed_line5: "";
speed_label5: "";
call_hold_ringback: "0";
dnd_control: "0";
anonymous_call_block: "0";
callerid_blocking: "0";
enable_vad: "0";
semi_attended_transfer: "1";
call_waiting: "1";
cfwd_url: "";
cnf_join_enable: "1";
phone_label: "";
preferred_codec: "g711ulaw";
```

Configuring Dial-Plan Patterns

To create and apply a sequence of digits that specifies a global prefix for expanding individual abbreviated SIP extensions into fully qualified E.164 numbers, follow the steps in this section.

Additional dial peers are built for the expanded numbers when the SIP phone registers in Cisco Unified CME. The **show voice register dial-peer** command displays all the dial peers created for SIP phones that have registered.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **dialplan-pattern** *tag pattern* **extension-length** *extension-length* [**extension-pattern** *extension-pattern*] [**no-reg**]
5. **call-forward system redirecting-expanded**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	`enable` **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	`configure terminal` **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	`voice register global` **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.

	Command or Action	Purpose
Step 4	**dialplan-pattern** *tag pattern* **extension-length** *extension-length* [**extension-pattern** *extension-pattern* | **no-reg**] **Example:** `Router(config-register-global)# dialplan-pattern 1 4085550... extension-length 4`	Creates a global prefix that can be used to expand extension numbers of incoming and outgoing calls to and from SIP phones. • *tag*—Identifies this dial-plan pattern. The tag is a number from 1 to 5. • *pattern*—Dial-plan pattern, such as area code, prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits. • **extension-length**—Sets the number of extension digits that will appear as a caller ID. • *extension-length*—Number of extension digits. The extension length must match the setting for IP phones. The range is from 1 to 32. • **extension-pattern**—(Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the *pattern* variable. • *extension-pattern*—(Optional) Leading digit pattern for extension. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599, and 5... would include 5000 to 5999. The extension pattern must match the setting for SIP phones in Cisco Unified CME mode. • **no-reg**—(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
Step 5	**call-forward system redirecting-expanded** **Example:** `Router(config-register-global)# call-forward system redirecting-expanded`	Applies dial-plan pattern expansion globally to redirecting number for SIP extensions in Cisco Unified CME for call forward using B2BUA. • **system**—Defines call-forward system parameter. • **redirecting-expanded**—Expands the redirecting extension to an E.164 number.
Step 6	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows a configuration for two Cisco Unified CME systems. Each is configured with the same **dialplan-pattern** commands, but one system uses 50.. and the other uses 60.. for extension numbers. Calls from the "50.." system to the "60.." system, and vice versa, are treated as internal calls. Calls that go across a H.323 network and calls that go to a PSTN through an ISDN interface on one of the configured Cisco Unified CME routers are represented as E.164.

```
Router(config)# voice register global
```

```
Router(config-register-global)# dialplan-pattern 1 40855550.. extension-length 4
extension-pattern 50..
Router(config-register-global)# dialplan-pattern 2 51055560.. extension-length 4
extension-pattern 60..
```

Applying Voice Translation Rules

To apply a preconfigured voice translation rule to modify the number dialed by extensions on a SIP phone, follow the steps in this section.

Translation rules perform regular expression matches and replace substrings of a called, calling, or redirecting numbers if the number matches the match pattern, number plan, and type defined in the rule.

Prerequisites

Translation rule to be applied must be configured using the **voice translation-rule** command. See the "Voice Translation Rules and Profiles" section in the *Cisco Unified CME System Administrator Guide* for your version at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **voice register pool** *tag*
5. **translate-outgoing** {**called** | **calling**} *rule-tag*
6. **end**

DETAILED STEPS

Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.

Step 4	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 5	**translate-outgoing** {**called** | **calling**} *rule-tag* **Example:** `Router(config-register-pool)# translate-outgoing called 1`	(Optional) Selects a preconfigured number translation rule to modify the number dialed by a specific extension. • **called**—Called party requires translation. • **calling**—Calling party requires translation. • *rule-tag*—Arbitrarily chosen number by which the rule set is referenced. The range is from 1 to 2147483
Step 6	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
```

What to Do Next

- If the following statements are true and your SIP phones are not already connected to your network, connect your SIP phones now. To find installation instructions for your Cisco Unified IP phone, select your phone type under the “IP Phone” heading at http://www.cisco.com/en/US/products/sw/voicesw/tsd_products_support_category_home.html. Then, see the“Verifying the Configuration” section on page 62.
  - The appropriate phone firmware, including all versions required during an upgrade sequence, are installed in flash memory of the TFTP server from which the Cisco Unified IP phones will download their configuration profiles.
  - You have configured the **upgrade** command to upgrade phone firmware as required to ensure that each Cisco Unified IP phone will download the appropriate phone firmware before registering.
  - You have verified the configuration profiles for your SIP phones.
- If the following statements are true and the SIP phones are already connected to Cisco Unified CME, see the “Resetting SIP Phones in Cisco Unified CME” section on page 60.
  - The appropriate phone firmware, including all versions required during an upgrade sequence, are installed in flash memory of the TFTP server from which the Cisco Unified IP phones will download their configuration profiles.

- You have configured the **upgrade** command to upgrade phone firmware as required to ensure that each Cisco Unified IP phone will download the appropriate phone firmware before registering.
- You have verified the configuration profiles for your SIP phones.

Resetting SIP Phones in Cisco Unified CME

To reset an individual or all SIP phones connected to Cisco Unified CME, use either the **reset (voice register pool)** or **reset (voice register global)** command.

Tip If phones are not yet plugged in, this task is not necessary. Instead, connect your IP phones to your network to boot the phone and download the required configuration files. To find installation instructions for your Cisco Unified IP phone, select your phone type under the "IP Phone" heading at http://www.cisco.com/en/US/products/sw/voicesw/tsd_products_support_category_home.html. Then, see the"Verifying the Configuration" section on page 62.

Using the reset Command to Reboot All SIP Phones

To reset all SIP phones after you make changes to any of the following configuration parameters, use the **reset** command in voice register global configuration mode.

- Source address
- Date and time settings
- Service URL
- TFTP path
- Voicemail access number

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **reset**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**reset** **Example:** `Router(config-register-global)# reset`	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router and contacts the DHCP server and the TFTP server for updated information.
Step 6	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Using the reset Command to Reset an Individual SIP Phone

To reset an individual SIP phone after you make changes to any of the following configuration parameters, use the **reset** command in voice register pool configuration mode.

- Codec
- Extension number
- Display name or label
- Username and password

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **exit**

6. **voice register pool** *pool-tag*
7. **reset**
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 6	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 7	**reset** **Example:** `Router(config-register-pool)# reset`	Performs a complete reboot of the single SIP phone specified with the **voice register pool** command and contacts the DHCP server and the TFTP server for updated information.
Step 8	**end** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Verifying the Configuration

Verify that Cisco IP phones in Cisco Unified CME can place and receive calls through the voice ports.

Step 1 Test local phone operation. Make calls between SIP phones on the Cisco Unified CME router

Step 2 Place a call from an IP phone in Cisco Unified CME to a number in the local calling area.

Step 3 Place a call to an IP phone in Cisco CME from a phone outside this Cisco Unified CME system.

Where to Go Next

After creating the basic configuration profiles for SIP phones in Cisco Unified CME and verifying local and external call operation, you are ready to configure additional Cisco Unified CME features for your SIP phones. See the "Configuring Cisco Unified CME Features for SIP Phones" section on page 65.

Configuring Cisco Unified CME Features for SIP Phones

First Published: June 19, 2006

This chapter describes how to set up features for all SIP phone users system wide, and also for individual SIP phones phones in a Cisco Unified CallManager Express (Cisco Unified CME) system

Finding Feature Information

Your Cisco IOS software release may not support all of the features documented in this guide. To reach links to specific feature documentation in this guide and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Unified CME for SIP Phones" section in this guide.

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Finding Support Information for Cisco Unified CME

For information about Cisco IOS software and Cisco Unified CME compatibility, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

For information about Cisco Unified CME specifications, including number of supported phones, see the appropriate *Cisco Unified CME Firmware, Platforms, Memory, and Voice Products* document at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Contents

How to Configure Cisco Unified CME Features for SIP Phones

This section includes the following tasks:

Configuring After Hours Call Blocking

Follow the steps in this section to perform the following tasks:

- Configure the block criteria for all SIP phones in Cisco Unified CME.
- Exempt individual SIP phones or directory numbers associated with a SIP phone from the call-blocking criteria.

Cisco Unified CME provides the same time-based call-blocking mechanism that is currently provided for SCCP phones and expands it to SIP endpoints. Call blocking to prevent unauthorized use of Cisco Unified IP phones is implemented by matching a pattern of specified digits during a particular time of the day and day of the week or date.You can specify up to 32 patterns of digits for blocking. This feature supports incoming SIP and analog FXS calls. The “Login” toll-bar override is not supported.

The Cisco Unified CME session application accesses the current after-hours configuration and applies it to calls originated by SIP phones that are registered to the Cisco Unified CME router. The after-hours commands are the same as for SCCP phones in Cisco Unified CME.

When a user attempts to place a call to digits that match a pattern that has been specified for call blocking during a time period that has been defined for call blocking, the call is immediately terminated and the caller will hear a fast busy signal.

The after-hours configuration applies globally to all dial peers in Cisco Unified CME. There is no pin to bypass blocking from SIP phones and you can disable the feature for an individual extension on a SIP phone or for a particular voice pool.

The **show voice register dial-peer** command displays all the dial peers created dynamically by SIP phones that have registered. This command also displays configurations for after hours blocking and call forwarding.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony service**
4. **after-hours block pattern** *pattern-tag pattern* [*7-24*]
5. **after-hours date** *month date start-time stop-time*

6. **after-hours day** *day start-time stop-time*
7. **exit**
8. **voice register pool** *pool-tag*
   or
   voice register dn *dn-tag*
9. **after-hour exempt**
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony service** **Example:** `Router(config)# telephony service`	Enters telephony service configuration mode.
Step 4	**after-hours block pattern** *pattern-tag pattern* [**7-24**] **Example:** `Router(config-telephony)# after-hours block pattern 2 91`	Defines patterns for outgoing calls. • *pattern-tag*—Unique number pattern for call blocking. Define up to 32 call-blocking patterns in separate commands. Range is 1 to 32. • *pattern*—Specifies outgoing call digits to be matched for blocking. • (Optional) The **7-24** keyword specifies that the pattern always be blocked, 7 days a week, 24 hours a day. Otherwise, the pattern is blocked during the days and dates that are defined with the **after-hours day** and **after-hours date** commands.

	Command or Action	Purpose
Step 5	**after-hours date** *month date start-time stop-time* **Example:** `Router(config-telephony)# after-hours date jan 1 0:00 23:59`	Defines a recurring period based on date of month during which outgoing calls that match defined block patterns are blocked on IP phones. • *month*—Valid abbreviations are **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**. • *date*—Range is 1 to 31. • Enter beginning and ending times for call blocking in an HH:MM format using a 24-hour clock. The *stop-time* must be greater than the *start-time*. The value 24:00 is not valid. If you enter 00:00as a stop time, it is changed to 23:59. If you enter 00:00 for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.
Step 6	**after-hours day** *day start-time stop-time* **Example:** `Router(config-telephony)# after-hours day sun 0:00 23:59`	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones • *day*—Valid abbreviations are **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**, **sun**. • Enter beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The *stop-time* must be greater than the *start-time*. The value 24:00 is not valid. If you enter 00:00 as a stop time, it is changed to 23:59. If you enter 00:00 for both start time and stop time, calls are blocked for the entire 24-hour period on the specified day.
Step 7	**exit** **Example:** `Router(config-telephony)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 8	**voice register pool** *pool-tag* or **voice register dn** *dn-tag* **Example:** `Router(config)# voice register pool 1` or `Router(config)# voice register dn 1`	Enters voice register pool configuration mode to set parameters for specified SIP phone. • *pool-tag* unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command. or Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number during configuration tasks. Range is 1 to 150 or the maximum defined by the **max-dn (voice register global)** command.

	Command or Action	Purpose
Step 9	**after-hour exempt** **Example:** `Router(config-register-pool)# after-hour exempt` or `Router(config-register-dn)# after-hour exempt`	Exempts all numbers on a SIP phone from call blocking. or Exempts an individual directory number from call blocking.
Step 10	**end** **Example:** `Router(config-register-dn)# end` or `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to define two sets of call-blocking criteria. Pattern 1 blocks any outgoing call that requires the user to dial "9011," 7 days a week, 24 hours a day. Pattern 2 blocks any outgoing call that requires the user to dial "91," from 0:00 AM to 11:59 PM on Sundays and on Jan. 1 only.

```
telephony-service
 after-hours block pattern 1 9011 7-24
 after-hours block pattern 2 91
 after-hours block Sun 0:00 23:59
 after-hours block Jan 1 0:00 23:59
```

The following examples show how to exempt either an individual directory number or all numbers on an individual SIP phone, from the configured call-blocking criteria:

```
voice register dn 1
 after-hour exempt
```

or

```
voice register pool 1
 after-hour exempt
```

Configuring SIP-to-SIP Phone Call Forwarding

To configure SIP-to-SIP call forwarding using a back-to-back user agent (B2BUA), which allows call forwarding on any dial peer, follow the steps in this section.

The Cisco Unified CME acts as both UA server and UA client; that is, as a B2BUA. Calls into a SIP phone can be forwarded to other SIP or SCCP devices (including Cisco Unity, third-party voice mail systems, an auto attendant or an IVR system, such as Cisco Unified IPCC and Cisco Unified IPCC Express). In addition, SCCP phones can be forwarded to SIP phones.

Cisco Unity or other voice-messaging systems connected by a SIP trunk or SIP user agent are able to pass an MWI to a SIP phone when a call is forwarded. The SIP phone then displays the MWI when indicated by the voice-messaging system.

The call-forward busy response is triggered when a call is sent to a SIP phone using a VoIP dial peer and a busy response is received back from the phone. SIP-to-SIP call forwarding is invoked only if the phone is dialed directly. Call forwarding is not invoked when the phone number is called through a sequential, longest-idle, or peer hunt group.

Note You can configure call forwarding for an individual extension on a SIP phone, or to the number on which the SIP phone appears. If the information is configured in both, the information under voice register dn takes precedence over the information configured under voice register pool

Prerequisites

For SIP-to-SIP call forwarding, connections between specific types of endpoints in a Cisco IP-to-IP gateway must be configured by using the **allow-connections** command. For additional information, see the "Enabling SIP-to-SIP Calls in Your VoIP Network" section on page 35.

Restrictions

- SIP-to-SIP call forwarding is invoked only if that phone is dialed directly. Call forwarding is not invoked when the phone number is called through a sequential, longest-idle, or peer hunt group.
- If call forwarding is configured for a hunt group member, call forward is ignored by the hunt group.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
   or
   voice register dn *dn-tag*
4. **call-forward b2bua all** *directory-number*
5. **call-forward b2bua busy** *directory-number*
6. **call-forward b2bua mailbox** *directory-number*
7. **call-forward b2bua noan** *directory-number* **timeout** *seconds*
8. **call-forward b2bua unreachable** *directory-number*
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* or **voice register dn** *dn-tag* **Example:** `Router(config)# voice register pool 1` or `Router(config)# voice register dn 1`	Enters voice register pool configuration mode to set parameters for specified SIP phone. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool (voice register global)** command. or Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number (extension) during configuration tasks. Range is 1 to 150 or the maximum defined by the **max-dn (voice register global)** command.
Step 4	**call-forward b2bua all** *directory- number* **Example:** `Router(config-register-pool)# call-forward b2bua all 5005` or `Router(config-register-dn)# call-forward b2bua all 5005`	Enables call forwarding for a SIP back-to-back user agent so that all incoming calls will be forwarded to the designated directory-number. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number.
Step 5	**call-forward b2bua busy** *directory- number* **Example:** `Router(config-register-pool)# call-forward b2bua busy 5006` or `Router(config-register-dn)# call-forward b2bua busy 5006`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that is busy will be forwarded to the designated directory number. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number.

	Command or Action	Purpose
Step 6	**call-forward b2bua mailbox** *directory- number* **Example:** `Router(config-register-pool)# call-forward b2bua mailbox 5007` or `Router(config-register-dn)# call-forward b2bua mailbox 5007`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls that have been forwarded to a busy or no-answer extension will be forwarded to the recipient's voice mail. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number.
Step 7	**call-forward b2bua noan** *directory- number* **timeout** *seconds* **Example:** `Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10` or `Router(config-register-dn)# call-forward b2bua noan 5010 timeout 10`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that does not answer will be forwarded to the designated directory number. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number. • **timeout** *seconds*—Duration that a call can ring before it is forwarded to the destination directory number. Range is 3 to 60000, The default value is 20
Step 8	**call-forward b2bua unreachable** *directory- number* **Example:** `Router(config-register-pool)# call-forward b2bua unreachable 5009` or `Router(config-register-dn)# call-forward b2bua unreachable 5009`	Enables call forwarding for a SIP back-to-back user agent so that calls can be forwarded to a phone that has not registered in Cisco Unified CME. The designated directory-number must be configured in Cisco Unified CME. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number.
Step 9	**end** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Configuring Call Transfer

Call-transfer features are applied to individual SIP phones in Cisco Unified CME by means of templates that are created to enable common parameters. See the "Creating and Applying Templates to SIP Phones" section on page 88.

Configuring Call Waiting Beep

To set a repeating audible alert notification system wide to indicate when a call is on hold on a SIP phone in Cisco Unified CME, use the **hold-alert** command.

Note The Call Waiting feature is enabled by default; the Call Waiting feature can be disable or enabled for an individual SIP phone in Cisco Unified CME using the **call-waiting (voice register pool)** command. See the "Specifying Optional Phone-Specific Parameters for SIP Phones in Cisco Unified CME" section on page 48.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **hold-alert** *timeout*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**hold-alert** *timeout* **Example:** `Router(config-register-global)# hold-alert 30`	Sets an audible alert notification when a call is on hold on a SIP phone. Default is disabled. • *timeout*—Interval after which an audible alert notification is repeated, in seconds. Range is from 15 to 300.
Step 6	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to set the hold alert to 30 seconds on SIP phones in Cisco Unified CME.

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# hold-alert 30
```

Configuring a Distinctive Ring

To configure a distinctive ring for SIP phones in Cisco Unified CME, follow the steps in this section.

Distinctive ring is used to identify internal and external calls. An internal calls is defined as a call originating from any Cisco SIP or Cisco SCCP IP phone that is registered in Cisco Unified CME or is routed through the local FXS port.

The type of ring sound requested is signaled to the SIP phone using an alert-info signal. If this feature is enabled, Cisco Unified CME generates the alert-info for incoming calls from any phone that is not registered in Cisco Unified CME, to the local SIP endpoint. Alert-info from an incoming SIP leg can be relayed to an outgoing SIP leg with the internally generated alert-info taking precedence.
Cisco Unified IP phones use the standard Telecordia Technologies distinctive ring types.

Note bellcore-dr1 to bellcore-dr5 are the only Telecordia options that are supported for SIP phones.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **external-ring** {**bellcore-dr1** | **bellcore-dr2** | **bellcore-dr3** | **bellcore-dr4** | **bellcore-dr5**}
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice register global`** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.

	Command or Action	Purpose
Step 4	**external-ring** {**bellcore-dr1** | **bellcore-dr2** | **bellcore-dr3** | **bellcore-dr4** | **bellcore-dr5**} **Example:** `Router(config-register-global)# external-ring bellcore-dr3`	Specifies the type of ring sound used on SIP or SCCP phones for external calls • Default—Internal ring sound is used for all incoming calls.
Step 5	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Configuring Voice Hunt Groups

To apply the hunt-group feature to a list of SIP extensions and redirect calls for a specific number (hunt-group pilot number) to a defined group of directory numbers, follow the steps in this section.

The **voice hunt-group** command uses the same shunt strategies and provides similar services as the **ephone-hunt** command for SCCP ephone-dn dial peers. SIP hunt-group types include sequential hunting by using fixed-order, round-robin circular, and longest-idle circular selection. This feature works with the B-ACD call queuing TCL scripts. Parallel hunt group (aka call blasting) for SIP VoIP dial peers is also supported in Cisco Unified CME. Call blasting is the back-to-back user agent version of call forking.

Restrictions

- SIP-to-H.323 calls are not supported.
- If call forward is configured for a hunt-group member, call forward is ignored by the hunt group.
- Forwarding or transferring to a voice hunt-group is not supported.
- Voice-class with codec list can be configured under voice register pool, and more than one list member will not be supported for B2BUA call.
- Caller ID updated is not supported for supplementary service.
- 100 voice hunt groups is the maximum numbers supported.
- Voice hunt groups are subject to max-redirect restriction.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice hunt-group** *hunt-tag* [**longest-idle** | **parallel** | **peer** | **sequential**]
4. **pilot** *number* [**secondary** *number*]
5. **list** *dn-number*, *dn-number*[, *dn-number...*]
6. **final** *final-number*
7. **preference** *preference-order* [**secondary** *secondary-order*]
8. **hops** *number*

9. **timeout** *seconds*
10. **default** *default-value*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice hunt-group`** *`hunt-tag`* `[`**`longest-idle`** `|` **`parallel`** `|` **`peer`** `|` **`sequential`**`]` **Example:** `Router(config)# voice hunt-group 1 longest-idle`	Enters voice hunt-group configuration mode to define a hunt group. • *hunt-tag*—Unique sequence number of the hunt group to be configured. Range is 1 to 100. • To define a new hunt group, enter one of the following **keywords**: – **longest idle**—Hunt group in which calls go to the directory number that has been idle for the longest time. – **parallel**—Hunt group in which calls simultaneously ring multiple phones. – **peer**—Hunt group in which the first directory number is selected round-robin from the list. – **sequential**—Hunt group in which directory numbers ring in the order in which they are listed, left to right. • The keyword is not required to change the hunt-group type. Remove the existing hunt group first by using the **no** form of the command; then, recreate the group.

	Command or Action	Purpose
Step 4	**pilot** *number* [**secondary** *number*] **Example:** `Router(config-voice-hunt-group)# pilot number 8100`	Defines the telephone number that callers dial to reach a voice hunt group. To remove the pilot number from the voice hunt group, use the **no** form of this command. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string may contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads. • **secondary** *number*—(Optional) Keyword and argument combination defines the number that follows as an additional pilot number for the voice hunt group. • Secondary numbers can contain wild cards in the string. A wildcard is a period (.), which matches any entered digit.
Step 5	**list** *directory-number*, *directory-number* [,*directory-number*...] **Example:** `Router(config-voice-hunt-group)# list 8000, 8010, 8020, 8030`	Creates a list of extensions that are members of a voice hunt group. To remove a list from a router configuration, use the **no** form of this command. • *directory-numbers*—List of extensions to be added as members to the voice hunt group. Separate the extensions with commas. • Add or delete all extensions in a hunt-group list at one time. You cannot add or delete a single number in an existing list. • There must be from 2 to 10 extensions in the hunt-group list, and each number must be a primary or secondary number. • A number cannot be added to a list unless it was already defined by using the **number (voice register dn)** or **number (and voice register pool)** command. • Any number in the list cannot be a pilot number of a parallel hunt group.
Step 6	**final** d*irectory-number* **Example:** `Router(config-voice-hunt-group)# final 8888`	Defines the last extension in a voice hunt group. • If a final number in one hunt group is configured as a pilot number of another hunt group, the pilot number of the first hunt group cannot be configured as a final number in any other hunt group.

	Command or Action	Purpose
Step 7	**preference** *preference-order* [**secondary** *secondary-order*] **Example:** `Router(config-voice-hunt-group)# preference 6`	Sets the preference order for the directory number associated with a voice hunt-group pilot number. **Note** It is recommended that the parallel hunt-group pilot number be unique in the system. Parallel hunt groups may not work if there are more than one partial or exact dial-peer match. For example, this happens if the pilot number is "8000" and there is another dial peer that matches "8…". If multiple matches cannot be avoided, give call parallel hunt group the highest priority to run by assigning a lower preference to the other dial peers. Note that 10 is the lowest preference value. By default, dial peers created by parallel hunt groups have a preference of 0. • *preference-order*—Range is from 0 to 8, where 0 is the highest preference, and 8 is the lowest preference. Default is 0. • **secondary** *secondary-order*—(Optional) Keyword and argument combination is used to set the preference order for the secondary pilot number. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference. Default is 9.
Step 8	**hops** *number* **Example:** `Router(config-voice-hunt-group)# hops 2`	For configuring a peer or longest-idle voice hunt group only. Defines the number of times that a call can hop to the next number in a peer or longest-idle voice hunt group before the call proceeds to the final number. • *number*—Number of hops. Range is 2 to 10, and the value must be less than or equal to the number of extensions specified by the **list** command. • Default is the same number as there are destinations defined under the **list** command.
Step 9	**timeout** *seconds* **Example:** `Router(config-voice-hunt-group)# timeout 100`	Defines the number of seconds after which a call that is not answered is redirected to the next directory number in a voice hunt-group list. • Default is 180 seconds.

	Command or Action	Purpose
Step 10	**default** *default-value* **Example:** `Router(config-voice-hunt-group)# default timeout`	Configures the default value for a specific voice hunt group command. • A separate **default** command is required for each voice hunt group command to be configured. • *default-value*—One of the voice hunt group configuration commands. Valid choices are as follows: – hops (Peer or longest-idle voice hunt group only) – preference – timeout
Step 11	**end** **Example:** `Router(config-voice-hunt-group)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following is an example of a longest-idle voice hunt group:

```
voice hunt-group 1 longest-idle
 pilot 8100
 list 8000, 8010, 8020, 8030
 final 8888
 timeout 10
```

The following is an example of a parallel voice hunt group:

```
voice hunt-group 2 parallel
 pilot 6000
 list 3000, 3010, 3020
 final 9999
 timeout 10
```

The following is an example of a peer voice hunt group:

```
voice hunt-group 11 peer
 pilot 7000
 list 7001, 7010, 7020
 final 7777
 timeout 20
```

The following is an example of a sequential voice hunt group:

```
voice hunt-group 31 sequential
 pilot 3100
 list 3130, 3136, 3100
 final 3131
 timeout 20
```

Configuring Huntstop

To configure the Huntstop feature and prevent hunt-on-busy from redirecting a call from a busy phone into a dial peer that has been setup with a catch-all default destination, follow these steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number** *number*
5. **huntstop**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register dn** *dn-tag* **Example:** `Router(config-register-global)# voice register dn 1`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number (extension) during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn (voice register global)** command.
Step 4	**number** *number* **Example:** `Router(config-register-dn)# number 5001`	Defines a valid number for a directory number. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string may contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.
Step 5	**huntstop** **Example:** `Router(config-register-dn)# huntstop`	Disables call-hunting behavior for an extension on a SIP phone. • Default is huntstop is disabled.
Step 6	**end** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows a typical configuration in which huntstop is required. The **huntstop** command is enabled and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy (three periods are used as wild cards).

```
voice register dn 1
 number 5001
 huntstop

voice register pool 4
 number 1 dn 1
 id-mac 0030.94c3.8724

dial-peer voice 5000 voip
 destination-pattern 5...
 session target ipv4:192.168.17.225
 session protocol sipv2
```

Configuring Bulk Registration

To configure bulk registration for registering a block of phone numbers with an external registrar so that calls can be routed to Cisco Unified CME from the SIP network, follow the steps in this section.

Numbers that match the defined number pattern defined can register with the external registrar. The block of numbers that is registered can include any phone that is attached to Cisco Unified CME using SIP or SCCP, or any analog phone that is directly attached to a Cisco router FXS port.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **bulk** *number*
6. **authenticate** [**all**][**realm** *string*]
7. **exit**
8. **voice register pool** *pool-tag*
9. **username** user*name* **password** *string*
10. **exit**
11. **sip-ua**
12. **registrar** {**dns**:*address* | **ipv4**:*destination-address*} **expires** *seconds* [**tcp**] [**secondary**] **no registrar** [**secondary**]
13. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice register global`** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**`mode`** *`cme`* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**`bulk`** *`number`* **Example:** `Router(config-register-global)# bulk 408526....`	Sets bulk registration for E.164 numbers that will register with SIP proxy server. • *number*—Unique sequence of up to 32 characters including wild cards and patterns that represents E.164 n umbers that will register with Sip proxy server.
Step 6	**`authenticate`** `[`**`all`**`][`**`realm`** *`string`*`]` **Example:** `Router(config-register-global)# authenticate all realm company.com`	(Optional) Enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods. Required only if you want to enable authentication. • **realm** *string*—(Optional) Keyword and argument combination for challenge and response as specified in RFC 2617.
Step 7	**`exit`** **Example:** `Router(config-register-pool)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 8	**`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 3`	Required only if authentication is enabled with the **authenticate (voice register global)** command. Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.

	Command or Action	Purpose
Step 9	**username** *username* **password** *string* **Example:** `Router(config-register-pool)# username smith password 123zyx`	Required only if authentication is enabled with the **authenticate (voice register global)** command. Creates an authentication credential for SIP phone registration only. **Note** This command is not for SIP proxy registration. The password will not be encrypted. All lines in a phone will share the same credential. • *username*—Identifies a local Cisco Unified IP phone user. Default is Admin. • **password**—Enables a password for the local Cisco Unified IP phone user. • Enter *password* string.
Step 10	**exit** **Example:** `Router(config-register-pool)# exit`	Required only if authentication is enabled with the **authenticate (voice register global)** command. Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 11	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters Session Initiation Protocol (SIP) user agent (ua) configuration mode for configuring the user agent.
Step 12	**registrar** {**dns**:*address* | **ipv4**:*destination-address*} **expires** seconds [**tcp**] [**secondary**] **no registrar** [**secondary**] **Example:** `Router(config-sip-ua)# registrar server ipv4:1.5.49.240`	Enables SIP gateways to register E.164 numbers with SIP proxy server. • **dns**:*address*—Specifies the DNS address of the primary SIP registrar server. • **ipv4**:*destination- address*—Specifies the IP address of the primary SIP registrar server. • **expires** *seconds*—(Optional) Default registration time, in seconds. Range is 60 to 65535. Default is 3600. • **tcp**—(Optional) Sets the transport layer protocol to TCP. UDP is the default. • **secondary**—(Optional) Specifies registration with a secondary SIP proxy or registrar to provide redundancy if the primary registrar fails.
Step 13	**end** **Example:** `Router(config-sip-ua)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows all phone numbers that match the pattern “408555..” registering with the SIP proxy server (IP address 1.5.49.240):

```
voice register global
 mode cme
 bulk 408555….
```

```
sip-ua
 registrar ipv4:1.5.49.240
```

What to Do Next

If you want to configure a particular directory number to not register with the external SIP proxy, see the "Disabling SIP Proxy Registration for a Particular Directory Number" section on page 84. Otherwise, you are ready to configure additional per-phone features for SIP phones. See the "Configuring Cisco Unified CME Phone Features for SIP Phones" section on page 87.

Disabling SIP Proxy Registration for a Particular Directory Number

To configure a particular directory number to not register with an external SIP proxy server, follow the steps in this task.

The **voice register dn** command is an alternate or supplement to bulk registration since it allows Cisco Unified CME to register an individual phone number or directory number to the external registrar. Phone numbers that are registered under voice register dn must belong to a SIP phone that is itself registered in Cisco Unified CME. You can specify that a particular directory number not register with the external registrar.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number** number
5. **no-reg**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**`voice register dn`** *`dn-tag`* **Example:** `Router(config-register-global)# voice register dn 1`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn (voice register global)** command.
Step 4	**`number`** *`number`* **Example:** `Router(config-register-dn)# number 4085550152`	Defines a valid number for a directory number to be assigned to a SIP phone in Cisco Unified CME. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string may contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.
Step 5	**`no-reg`** **Example:** `Router(config-register-dn)# no-reg`	Causes directory number specified by the **voice register-dn** command to not register with an external proxy server.
Step 6	**`end`** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows all phone numbers that match the pattern “408555..” registering with the SIP proxy server (IP address 1.5.49.240) *except* directory number 1, number “4085550101,” for which bulk registration is disabled with the **no-reg** command:

```
voice register global
 mode cme
 bulk 408555….
voice register dn 1
 number 4085550101
 no-reg
sip-ua
 registrar ipv4:1.5.49.240
```

Where to Go Next

- If you want to create or modify optional features that affect individual SIP phones in Cisco Unified CME, see the “Configuring Cisco Unified CME Phone Features for SIP Phones” section on page 87.
- If you want to configure support for voice-mail messaging for SIP phones connected directly in Cisco Unified CME, see the “Configuring Voice Mail Integration with Cisco Unified CME for SIP Phones” section on page 99.

- If you are finished creating or modifying Cisco Unified CME for SIP phones, you must reset or reboot the SIP phones connected directly in Cisco Unified CME. See the "Resetting SIP Phones in Cisco Unified CME" section on page 60.

Configuring Cisco Unified CME Phone Features for SIP Phones

First Published: June 19, 2006

This chapter describes how to configure optional features that affect individual SIP phones in Cisco Unified CallManager Express (Cisco Unified CME).

Finding Feature Information

Your Cisco IOS software release may not support all of the features documented in this guide. To reach links to specific feature documentation in this guide and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Unified CME for SIP Phones" section in this guide.

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Finding Support Information for Cisco Unified CME

For information about Cisco IOS software and Cisco Unified CME compatibility, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

For information about Cisco Unified CME specifications, including number of supported phones, see the appropriate *Cisco Unified CME Firmware, Platforms, Memory, and Voice Products* document at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Contents

How to Configure Cisco Unified CME Phone Features for SIP Phones

This section contains the following tasks:

- Creating and Applying Templates to SIP Phones, page 88
- Configuring Conference Call Features, page 90
- Configuring Do Not Disturb (DND), page 93
- Configuring Intercom Auto Answer, page 95
- Configuring Speed Dial Features, page 96

Creating and Applying Templates to SIP Phones

To create templates of common features and softkeys that can be applied to individual Cisco SIP IP phones, follow the steps in this section.

Use the **voice register template** command to create up to five different templates containing one or more phone system features and softkeys to expedite configuring individual SIP phones in Cisco Unified CME.

After creating a template, use the **template** command under the voice register pool configuration mode to apply the template to a particular SIP phone. Only one template can be applied to a SIP phone at a time. If you apply a second template to a SIP phone, the second template will overwrite the first template information, but the new information will only take affect after you use the **reset** command to reboot the phone. If you do not reboot the Cisco Unified IP phone, the previously configured template remains in effect.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **exit**
6. *command*
7. **voice register template** *template-tag*
8. **exit**
9. **voice register pool** *pool-tag*
10. **template** *template-tag*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 6	**voice register template** *template-tag* **Example:** `Router(config)# voice register template 1`	Enters voice register template configuration mode to define a template of common parameters for SIP phones in Cisco Unified CME. • Range is 1 to 5.
Step 7	*command* **Example:** `Router(config-register-template)# anonymous block`	Applies the specified command to this template and enables the corresponding feature on any supported SIP phone that uses a template in which this command is configure. • Type **?** to display list of commands that can be used in a voice register template. • Repeat this step for each feature to be added to this voice register template.
Step 8	**exit** **Example:** `Router(config-register-template)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 9	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the Cisco SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.

	Command or Action	Purpose
Step 10	**`template`** *`template-tag`* **Example:** `Router(config-register-pool)# voice register pool 1`	Applies a template created with the **voice register template** command. • *template-tag*—Unique sequence number of the template to be applied to the SIP phone specified by the **voice register pool** command. Range is 1 to 5.
Step 11	**`end`** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows templates 1 and 2 and how to do the following:

- Apply template 1 to SIP phones 1 to 3
- Apply template 2 to SIP phone 4
- Remove a previously created template 5 from SIP phone 5.

```
Router(config)# voice register template 1
 Router(config-register-temp)# anonymous block
 Router(config-register-temp)# caller-id block
 Router(config-register-temp)# voicemail 5001 timeout 15

Router(config)# voice register template 2
 Router(config-register-temp)# anonymous block
 Router(config-register-temp)# caller-id block
 Router(config-register-temp)# no conference
 Router(config-register-temp)# no transfer-attended
 Router(config-register-temp)# voicemail 5005 timeout 15

Router(config)# voice register pool 1
 Router(config-register-pool)# template 1

Router(config)# voice register pool 2
 Router(config-register-pool)# template 1

Router(config)# voice register pool 3
 Router(config-register-pool)# template 1

Router(config)# voice register pool 4
 Router(config-register-pool)# template 2

Router(config)# voice register pool 5
 Router(config-register-pool)# no template 5
```

Configuring Conference Call Features

To configure conference features for SIP phones in Cisco Unified CME, follow the steps in this section.

The maximum number of simultaneous conferences is platform specific to the type of Cisco Unified CME router, and each individual SIP phone can host a maximum of one conference at a time. You cannot create a second conference on a phone if you already have an existing conference on hold.

To adjust the gain level of an external call to provide more adequate volume, use the **max-conference** command. This functionality is applied to inbound audio packets so that conference participants can clearly hear a remote PSTN or VoIP caller joining the call. Note that this functionality cannot discriminate between a remote VoIP or foreign exchange office (FXO) source, which requires a volume gain, and a remote VoIP or Cisco Unified IP phone, which does not require a volume gain and may therefore incur some sound distortions.

A person who initiates a conference call and hangs up can either keep the remaining parties connected or disconnect them. The **keep-conference** command under voice register pool configures SIP phones to keep the remaining conference parties connected when the conference initiator hangs up (places the handset back in the on-hook position). Conference initiators can disconnect from the conference calls by pressing the Confrn (conference) softkey. Conference initiator drop-off can be configured per phone using the **conference (voice register template)** command.

When an initiator uses the Confrn key to disconnect from the conference call, the oldest call leg is placed on hold, leaving the initiator connected to the most recent call leg without consultation. The conference initiator can then navigate between the two parties by pressing either the Hold softkey or the line buttons to select the desired call. To facilitate call transfer, t**ransfer-attended (voice register template)** or the **transfer-blind (voice register template)** must be enabled.

Music on hold (MOH) is not supported for call hold invoked from a SIP phone. A caller hears only silence when placed on hold by a SIP phone.

Prerequisites

To facilitate call transfer by using the Confrn softkey, conference and transfer attended or transfer blind must be enabled.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony service**
4. **max-conferences** *max-conference-number* [**gain -6** | **0** | **3** | **6**]
5. **exit**
6. **voice register pool pool-tag**
7. **keep-conference**
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony service** **Example:** `Router(config)# telephony service`	Enters telephone-service configuration mode.
Step 4	**max-conferences** *max-conference-number* [**gain -6** | **0** | **3** | **6**] **Example:** `Router(config-telephony)# max-conferences 4`	Sets the maximum number of three-party conferences that are supported simultaneously by the Cisco Unified CME router. • Enter *max-conference-number*—Maximum number of three-party conferences that are supported simultaneously by a router. This number is platform-dependent, and the default is half the maximum for each platform. The following are the maximum values for this argument: – 8 for Cisco 1700 series, Cisco 2600 series, and Cisco 2801 routers. – 16 for Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3600 series, and Cisco 3700 series routers. – 24 (requires Cisco IOS Release 12.3(11)XL or higher) for Cisco 3800 series routers. • Each individual Cisco Unified IP phone can host only one conference at a time. You cannot create a second conference on the phone if you already have an existing conference on hold. • **gain**—(Optional) Increases the sound volume of VoIP and PSTN parties joining a conference call. The allowable decibel (db) units are -6, 0 db, 3, and 6. Default is -6 db.
Step 5	**exit** **Example:** `Router(config-telephony)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.

	Command or Action	Purpose
Step 6	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag* unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.
Step 7	**keep-conference** **Example:** `Router(config-register-pool)# keep-conference`	(Optional) Allows a Cisco Unified IP phone conference initiator to exit from conference calls and keeps the remaining parties connected. **Note** This step is included to illustrate how to enable the command if it was previously disabled. • Default is enabled. • Remaining calls are transferred without consultation as enabled by the **transfer-attended (voice register template)** or **transfer-blind (voice register template)** commands.
Step 8	**end** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Configuring Do Not Disturb (DND)

To configure the Do-not-Disturb (DND) feature set for Cisco Unified IP phones with softkeys, follow the steps in this section. When an incoming call is presented, the phone user sees two softkeys: Answer and DND.

The **call-forward b2bua noan** command forwards incoming calls to an extension that does not answer (NOAN) to another extension. If call-forward no-answer is configured, pressing the DND softkey will cause the an incoming call to be forwarded immediately to the specified number (typically a voice mail number).

If call-forward no-answer is not configured for the extension, pressing the DND key mutes the ringer until the call is terminated by the caller. The ringer mute action is temporary and applies only to the current call. The phone can be placed in a permanent muted ringer state by pressing the DND softkey while the phone is in an idle state. This permits incoming call screening because the phone display is still active and shows the caller ID for an incoming call.

To avoid all incoming calls, configure call-forward all-calls to set up unconditional forwarding of all incoming calls to voice mail or to another extension. For more information, see Configuring SIP-to-SIP Phone Call Forwarding in "Configuring Cisco Unified CME Features for SIP Phones" section on page 65.

The DND softkey can also be configured under the voice register template configuration mode. Commands under voice register template are used to create templates of common features and softkeys that can be applied to an individual phone. For more information, see "Creating and Applying Templates to SIP Phones" section on page 88.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **dnd**
5. **call-forward b2bua noan** *directory- number* **timeout** *seconds*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set parameters for specified SIP phone. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool (voice register global)** command.
Step 4	**`dnd`** **Example:** `Router(config-register-pool)# dnd-control`	Enables DND for the specified SIP phone being configured. **Note** If call forward no answer is not configured for the extension, pressing the DND key mutes the ringer until the caller terminates the call.
Step 5	**`call-forward b2bua noan`** *`directory- number`* **`timeout`** *`seconds`* **Example:** `Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10`	(Optional) Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that does not answer will be forwarded to the designated directory number. • *directory-number*—Up to 32 characters that represents a fully qualified E.164 telephone number. • **timeout** *seconds*—Duration that a call can ring before it is forwarded to the destination directory number. Range is 3 to 60000. The default value is 20.
Step 6	**`end`** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows the DND softkey enabled on the SIP phone that is identified as voice register pool 1:

```
Router# voice register pool 1
Router(config-register-pool)# dnd
```

Configuring Intercom Auto Answer

To enable the Intercom Auto Answer feature on a SIP phone extension, follow the steps in this section.

Use the **auto-answer (voice register dn)** command to create a Cisco Unified IP phone line connection that resembles a private line, automatic ring down (PLAR). Auto answer causes an extension to operate in auto-dial fashion for outbound calls and auto answer with mute for inbound calls. If an extension is configured for intercom operation, it can be associated with only one Cisco Unified IP phone.

Any caller can dial an intercom extension, and a call to an intercom extension that is originated by a non intercom caller is automatically answered, exactly like a legitimate intercom call. To prevent non intercom originators from manually dialing an intercom destination, use alphabetic characters when assigning numbers to intercom extensions by using the **number (voice register dn)** command. The alphabetic characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions whose calls are made by a Cisco Unified router.

Use the **reset** command to reset the Cisco Unified IP phone after configuring intercom auto answer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number** *number*
5. **auto-answer**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice register dn** *dn-tag* **Example:** `Router(config-register-global)# voice register dn 1`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number (extension) during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn (voice register global)** command.
Step 4	**number** *number* **Example:** `Router(config-register-dn)# number A5001`	Defines a valid number for a directory number to be assigned to a SIP phone in Cisco Unified CME. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string can contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.
Step 5	**auto-answer** **Example:** `Router(config-register-dn)# auto-answer`	Enables the Intercom Auto Answer feature on a SIP phone extension.
Step 6	**end** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to set the Intercom Auto Answer feature on SIP phone extension 1:

```
Router(config)# voice register dn 1
Router(config-register-dn)# number A5001
Router(config-register-dn)# auto-answer
```

Configuring Speed Dial Features

To configure Speed Dial features for Cisco SIP IP phones in Cisco Unified CME, use the **speed-dial (voice register pool)** command.

On Cisco Unified IP phones, speed-dial definitions are assigned to available buttons that have not been assigned to actual extensions. Speed-dial definitions are assigned in the order of their identifier numbers.

For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons have been assigned to extensions. If you used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone.

For Cisco Unified IP phones, speed-dial numbers can be assigned by the administrator and can be locked if the *digit-string* argument begins with a plus sign (+). Locked numbers cannot be changed at the phone.

Speed-dial instances without speed-dial numbers (those defined with only a pound sign) and speed-dial instances with unlocked *digit-string* arguments can be changed by users at their Cisco Unified IP phones.

Analog phone users who use a Cisco ATA-186 or Cisco ATA-188 to connect to Cisco Unified CME use a different method to access speed-dial numbers. Instead of pressing a speed-dial button, phone users with Cisco ATA devices press the asterisk (star) key and a *speed-tag* number (speed-dial identifier) to dial a speed-dial number. For instance, a phone user with a Cisco ATA-186 presses *1 to dial the number that has been programmed as speed-dial 1 on that phone. Phones with Cisco ATA devices are limited to a maximum of nine speed-dial numbers that must be programmed by the system administrator. The numbers cannot be programmed from the phone. With phones that use Cisco ATA devices, system administrators must be sure to tell phone users when speed-dial numbers have been programmed for their phones.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **speed-dial** *speed-tag digit-string* [**label** *label-text*]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 23`	Enters voice register pool configuration mode to set parameters for specified SIP phone. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool (voice register global)** command.

	Command or Action	Purpose
Step 4	**speed-dial** *speed-tag digit-string* [**label** *label-text*] **Example:** `router(config-register-pool)# speed-dial 2 +5001 label "Head Office"`	Creates a speed-dial definition for a SIP phone or analog phone that uses an analog adapter (ATA) in Cisco Unified CME. • *speed-tag*—Unique sequence number that identifies the speed-dial definition during configuration. Range is 1 to 5. • *digit-string*—Digits to be dialed when the speed-dial button is pressed on a SIP phone, or the digits to be dialed when the associated code is entered from an analog phone with a Cisco ATA device. **Note** For Cisco Unified IP phones, if the first character is the plus sign (+), this speed dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is configured. • **label** *label-text*—(Optional) Text string that is displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.
Step 5	**end** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to set speed-dial button 2 to dial the head office at extension 5001 and locks the setting so that the phone user cannot change the setting at the phone:

```
Router(config)# voice register pool 23
Router(config-register-pool)# speed-dial 2 +5001 label "Head Office"
```

Where to Go Next

After modifying or configuring optional features for individual SIP phones, you must reboot or reset the phones, see the "Resetting SIP Phones in Cisco Unified CME" section on page 60.

Configuring Voice Mail Integration with Cisco Unified CME for SIP Phones

First Published: June 19, 2006

This chapter describes how to configure support for voice-mail messaging for SIP phones connected directly in Cisco Unified CallManager Express (Cisco Unified CME).

Finding Feature Information

Your Cisco IOS software release may not support all of the features documented in this guide. To reach links to specific feature documentation in this guide and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Unified CME for SIP Phones" section in this guide.

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Finding Support Information for Cisco Unified CME

For information about Cisco IOS software and Cisco Unified CME compatibility, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

For information about Cisco Unified CME specifications, including number of supported phones, see the appropriate *Cisco Unified CME Firmware, Platforms, Memory, and Voice Products* document at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Contents

Prerequisites for Configuring Voice-Mail Integration with Cisco Unified CME for SIP Phones

- Voice mail must be installed and configured on your network.
- Calls can be placed between phones on the same Cisco Unified CME router.

How to Configure Voice-Mail Integration with Cisco Unified CME for SIP Phones

This section includes the following tasks:

Note For installation and configuration instructions for Cisco Unity, see the Install and Upgrade Guides for Cisco Unity at:
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/tsd_products_support_series_home.html.

For instructions on how to configure Cisco Unity Express, see the administrator guides in the Cisco Unity Express index at:
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/tsd_products_support_series_home.html.

For additional information about how to integrate Cisco Unified CME with Cisco Unity Express, see the *Integrating Documentation* at:
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Configuring a Voice Mailbox

To configure the telephone number that is speed-dialed when the Message button on a SIP phone is pressed, follow the steps in this section.

The same telephone number is configured for voice messaging for all the SIP phones in Cisco Unified CME. The **call forward b2bua** command enables call forwarding and designates that calls that are forwarded to a busy or no-answer extension be sent to a voicemail box.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **voicemail** *phone-number*
5. **exit**
6. **voice register dn** *dn-tag*
7. **number** *number*
8. **call-forward b2bua busy** *directory-number*
9. **call-forward b2bua mailbox** *directory-number*
10. **call-forward b2bua noan** *directory-number*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**voicemail** *phone-number* **Example:** `Router(config-register-global)# voice mail 1234`	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed. • *phone-number*—Same phone number is configured for voice messaging for all SIP phones in a Cisco Unified CME.
Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.

	Command or Action	Purpose
Step 6	**voice register dn** *dn-tag* **Example:** `Router(config-register-global)# voice register dn 2`	Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number (extension) during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn** command.
Step 7	**number** *number* **Example:** `Router(config-register-dn)# number 2200`	Defines a valid number for a directory number to be assigned to a SIP phone in Cisco Unified CME. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string can contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.
Step 8	**call-forward b2bua busy** *directory-number* **Example:** `Router(config-register-dn)# call-forward b2bua busy 1000`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that is busy will be forwarded to the designated directory number. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number.
Step 9	**call-forward b2bua mailbox** *directory-number* **Example:** `Router(config-register-dn)# call-forward b2bua mailbox 2200`	Designates voice mailbox to use at the end of a chain of call forwards. Incoming calls have been forwarded to a busy or no-answer extension will be forwarded to the directory-number specified. • *directory-number*— Up to 32 characters that represent a fully qualified E.164 telephone number.
Step 10	**call-forward b2bua noan** *directory-number* **timeout** *seconds* **Example:** `Router(config-register-dn)# call-forward b2bua noan 2201 timeout 15`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that does not answer will be forwarded to the designated directory number. • *directory-number*—Up to 32 characters that represent a fully qualified E.164 telephone number. • **timeout** *seconds*—Duration that a call can ring before it is forwarded to the destination directory number. Range is 3 to 60000. The default value is 20.
Step 11	**end** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to configure the call forward b2bua mailbox:

```
voice register global
 voicemail 1234
voice register dn 2
 number 2200
 call-forward b2bua all 1000
```

```
call-forward b2bua mailbox 2200
call-forward b2bua noan 2201 timeout 15
mwi
```

What to do Next

- To use a remote SIP-based IVR or Cisco Unity, or to connect Cisco Unified CME to a remote SIP-PSTN that goes through the PSTN to a voice-mail or IVR application, see the "Configuring a DTMF Relay Using SIP RFC 2833" section on page 103.
- To use voice mail on a a SIP network that connects to a Cisco Unity Express system, configure a nonstandard SIP NOTIFY format. See the "Configuring a DTMF Relay Using SIP NOTIFY (Nonstandard)" section on page 105.

Configuring a DTMF Relay Using SIP RFC 2833

To configure a SIP dial peer to point to Cisco Unity and enable SIP dual-tone multifrequency (DTMF) relay using RFC 2833, use the commands in this section on both the originating and terminating gateways.

SCCP phones provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voice-mail applications, Cisco Unified CME provides conversion from the out-of-band SCCP indication to the SIP standard for DTMF relay, which is RFC 2833. To select this method in the SIP VoIP dial peer, use the **dtmf-relay rtp-nte** command.

The SIP DTMF relay method is required in the following situations:

- When SIP is used to connect Cisco Unified CME to a remote SIP-based IVR or voice-mail application like Cisco Unity.
- When SIP is used to connect Cisco Unified CME to a remote SIP-PSTN voice gateway that goes through the PSTN to a voice-mail or IVR application.

Note The need to use out-of-band conversion is limited to SCCP phones. SIP phones natively support in-band DTMF relay as specified in RFC 2833.

Tip To configure a list of DTMF relay mechanisms for an individual SIP phone in Cisco Unified CME, see the "Specifying Phone-Specific Parameters for SIP Phones in Cisco Unified CME" section on page 45.

For information about the commands used during this task, see the *Cisco IOS Voice Command Reference* at
http://cisco.com/en/US/products/sw/iosswrel/ps5207/products_command_reference_book09186a0080 1a7f08.html.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **description** *string*

5. **destination-pattern** *string*
6. **session protocol sipv2**
7. **session target** {**dns**:*address* | **ipv4**:*destination-address*}
8. **dtmf-relay** [**rtp-nte**][**sip-notify**]
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router (config)# dial-peer voice 123 voip`	Enters dial-peer configuration mode to define a VoIP dial peer for the voice-mail system. • *tag*—Defines the dial peer being configured. Range is from 1 to 2147483647.
Step 4	**description** *string* **Example:** `Router (config-voice-dial-peer)# description CU pilot`	(Optional) Associates a description with the dial peer being configured. Enter a string of up to 64 characters.
Step 5	**destination-pattern** *string* **Example:** `Router (config-voice-dial-peer)# destination-pattern 20`	Specifies the pattern of the numbers that the user must dial to place a call. • *string*—Prefix or full E.164 number.
Step 6	session protocol sipv2 **Example:** `Router (config-voice-dial-peer)# session protocol sipv2`	Specifies a protocol for calls between local and remote routers using the packet network. • **sipv2**—Specifies SIP.
Step 7	**session target** {**dns**:*address* | **ipv4**:*destination-address*} **Example:** `Router (config-voice-dial-peer)# session target ipv4:10.8.17.42`	Designates a network-specific address to receive calls from the dial peer being configured. • **dns**:*address*—Specifies the DNS address of the voice-mail system. • **ipv4**:*destination- address*—Specifies the IP address of voice-mail system.

	Command or Action	Purpose
Step 8	**dtmf-relay** [**rtp-nte**][**sip-notify**] **Example:** `Router (config-voice-dial-peer)# dtmf-relay rtp-nte`	Sets DTMF relay method for the voice dial peer being configured. • **rtp-nte**—(Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
Step 9	**end** **Example:** `Router(config-voice-dial-peer)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows the configuration for a DTMF Relay:

```
dial-peer voice 1 voip
 destination-pattern 4…
 session target ipv4:10.8.17.42
 session protocol sipv2
 dtmf-relay sip-notify rtp-nte
```

What to Do Next

After integrating configuring DTMF relay, you are ready to configure Message Waiting Indicator (MWI) notification for either the MWI outcall, unsolicited notify, or subscribe/notify mechanism. See the "Configuring MWI" section on page 107

Configuring a DTMF Relay Using SIP NOTIFY (Nonstandard)

To configure a SIP dial peer to point to Cisco Unity Express and enable SIP dual-tone multifrequency (DTMF) relay using SIP NOTIFY format, follow the steps in this task.

Tip To configure a list of DTMF relay mechanisms for an individual SIP phone in Cisco Unified CME, see the "Specifying Optional Phone-Specific Parameters for SIP Phones in Cisco Unified CME" section on page 48.

For information about the commands used during this task, see the *Cisco IOS Voice Command Reference* at http://cisco.com/en/US/products/sw/iosswrel/ps5207/products_command_reference_book09186a00801a7f08.html.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **description** *string*

5. **destination-pattern** *string*
6. **b2bua**
7. **session protocol sipv2**
8. **session target** {**dns**:*address* | **ipv4**:*destination-address*}
9. **dtmf-relay** [**rtp-nte**][**sip-notify**]
10. **codec** *g711ulaw*
11. **no vad**
12. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal#`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router (config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode to define a VoIP dial peer for the voice-mail system. • *tag*—Defines the dial peer being configured. Range is from 1 to 2147483647.
Step 4	**description** *string* **Example:** `Router (config-voice-dial-peer)# description cue pilot`	(Optional) Associates a description with the dial peer being configured. Enter a string of up to 64 characters.
Step 5	**destination-pattern** *string* **Example:** `Router (config-voice-dial-peer)# destination-pattern 20`	Specifies the pattern of the numbers that the user must dial to place a call. • *string*—Prefix or full E.164 number.
Step 6	**b2bua** **Example:** `Router (config-voice-dial-peer)# b2bua`	Includes the Cisco Unified CME address as part of contact in 3XX response to point to Cisco Unity Express and enables SIP-to-SCCP call forward.
Step 7	**session protocol sipv2** **Example:** `Router (config-voice-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the packet network. • Enter **sipv2** to specify SIP.

	Command or Action	Purpose
Step 8	**session target** {**dns**:*address* | **ipv4**:*destination-address*} **Example:** `Router (config-voice-dial-peer)# ipv4:10.5.49.80`	Designates a network-specific address to receive calls from the dial peer being configured. • **dns**:*address*—Specifies the DNS address of voice-mail system. • **ipv4**:*destination- address*—Specifies the IP address of voice-mail system.
Step 9	**dtmf-relay** [**rtp-nte**][**sip-notify**] **Example:** `Router (config-voice-dial-peer)# dtmf-relay sip-notify`	Sets DTMF relay method for the voice dial peer being configured. • **sip-notify**—(Optional) Forwards DTMF tones using SIP NOTIFY messages.
Step 10	**codec** *g711ulaw* **Example:** `Router (config-voice-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for a dial peer being configured.
Step 11	**no vad** **Example:** `Router (config-voice-dial-peer)# no vad`	Disable voice activity detection (VAD) for the calls using the dial peer being configured.
Step 12	**end** **Example:** `Router(config-voice-dial-peer)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows the configuration for a DTMF relay:

```
dial-peer voice 1 voip
 destination-pattern 4…
 session target ipv4:10.5.49.80
 session protocol sipv2
 dtmf-relay sip-notify
 b2bua
```

What to Do Next

After configuring DTMF relay, you are ready to configure Message Waiting Indicator (MWI) notification for either the MWI outcall, unsolicited notify, or subscribe/notify mechanism. See the "Configuring MWI" section on page 107

Configuring MWI

Perform one of the tasks in this section, depending on whether you want to configure MWI outcall, unsolicited notify, or subscribe/notify for SIP endpoints in Cisco Unified CME.

• Defining MWI Outcall, page 108

- Configuring MWI Unsolicited Notify, page 110
- Configuring MWI Subscribe/Notify Server, page 112

Use the **mwi-server** command to request that the UA subscribe to a voice-mail server requesting notification of mailbox status. When there is a status change, the voice-mail server notifies the UA. The UA will indicate to the user that there is a change in mailbox status with an MWI tone when the user takes the phone off-hook. The **sip-server** and **mwi expires commands** under the telephony-service configuration mode have been migrated to **mwi-server** to support DNS format of the SIP server.

Configure the **mwi (voice register dn)** command to enable a phone extension, specified by the **voice register dn** command, to receive MWI notification using a SIP phone.

Note For extensions associated with analog telephone adaptors, the MWI is a lit function button on the Cisco ATA device and a stutter dial tone on the connected analog phone.

Defining MWI Outcall

To define a pilot call back number for MWI outcall, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag*
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. **mwi** {**off** | **on** | **on-off**}
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone-dn** *dn-tag* **Example:** `Router(config)# ephone dn 1`	Enters ephone-dn configuration mode and creates an ephone-dn for a Cisco Unified IP phone. • *dn-tag*—Unique sequence number that identifies this ephone-dn during configuration tasks. The maximum number of ephone-dns in Cisco Unified CME is version and platform specific. Type **?** to display the value.

	Command or Action	Purpose
Step 4	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 9000`	Associates a telephone or extension number with an ephone-dn. • *number*—String of up to 16 characters that represents an E.164 telephone number. Normally, the string is composed of digits, but the string can contain alphabetic characters when the number is dialed only by a router, as with an intercom number. One or more periods (.) can be used as wildcard characters. • **secondary**—(Optional) Associates the number that follows as an additional number for this ephone-dn. • **no-reg**—(Optional) Specifies that E.164 numbers in the dial peer do not register with the gatekeeper. • **both**—(Optional) Neither the primary nor secondary numbers will register with the gatekeeper. • **primary**—(Optional) Primary number will not register with the gatekeeper.
Step 5	**mwi** {**off** | **on** | **on-off**} **Example:** `Router(config-ephone-dn)# mwi on-off`	Enables this ephone-dn to receive message-waiting indication (MWI) notification from an external voice-messaging system. • **off**—Sets a Cisco Unified IP phone extension to process MWI to OFF, using either the main or secondary phone number. • **on**—Sets a Cisco Unified IP phone extension to process MWI to ON, using either the main or secondary phone number. • **on-off**—Sets a Cisco Unified IP phone extension to process MWI to both ON and OFF, using either the main or secondary phone number.
Step 6	**end** **Example:** `Router(config-ephone-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows an MWI callback pilot number:

```
ephone-dn 1
number 9000….
mwi on-off
```

What to Do Next

After integrating Cisco Unified CME and your voice mail system, you are ready to reboot or reset the phones. See the "Resetting SIP Phones in Cisco Unified CME" section on page 60.

Configuring MWI Unsolicited Notify

To identify the MWI server and specify a directory number to be enabled to receive MWI notification, follow the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*}
5. **exit**
6. **voice register dn** *dn-tag*
7. **number** *number*
8. **mwi**
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`sip-ua`** **Example:** `Router(config)# sip-ua`	Enters Session Initiation Protocol (SIP) user agent (ua) configuration mode for configuring the user agent.

	Command or Action	Purpose
Step 4	**mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*} [**expires** *seconds*] [**port** *port*] [**transport** {**tcp** | **udp**}] [**unsolicited**] **Example:** `Router(config-sip-ua)# mwi-server dns:server.yourcompany.com expires 60 port 5060 transport udp unsolicited`	Requests that the user-agent (UA) subscribe to a voice-mail server requesting notification of mailbox status. • **ipv4**:*destinatio-address* or **dns**: *host-name*—Keyword and argument combination to identify the mail server. • **expires** *seconds*—(Optional) Subscription expiration time, in seconds. The range is 1 to 999999. The default is 3600 • **port** *port*—(Optional) Defines the port number on the voice-mail server. The default is 5060. • **transport** {**tcp** | **udp**}—(Optional) Defines the transport protocol to the voice-mail server. UDP is the default • **unsolicited**—(Optional) Requires the voice-mail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.
Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 6	**voice register dn** *dn-tag* **Example:** `Router(config-register-global)# voice register dn 1`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn (voice register global)** command.
Step 7	**number** *number* **Example:** `Router(config-register-dn)# number 1234`	Defines a valid number for a directory number to be assigned to a SIP phone in Cisco Unified CME. • *number*—String of up to 16 characters that represents an E.164 telephone number. • Number string can contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.

	Command or Action	Purpose
Step 8	**mwi** **Example:** `Router(config-register-dn)# mwi`	Enables a specific directory number to receive MWI notification.
Step 9	**end** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to specify voice-mail server settings on a UA. The example includes the unsolicited keyword, enabling the voice-mail server to send a SIP notification message to the UA if the mailbox status changes and specifies that voice dn 1, number 1234 on the SIP phone in Cisco Unified CME will receive the MWI notification:

```
sip-ua
 mwi-server dns:server.yourcompany.com expires 60 port 5060 transport udp unsolicited
voice register dn 1
 number 1234
 mwi
```

What to Do Next

After integrating Cisco Unified CME and your voice mail system, you are ready to reboot or reset the phones. See the "Resetting SIP Phones in Cisco Unified CME" section on page 60.

Configuring MWI Subscribe/Notify Server

To define the MWI server and specify an directory number to be enabled to receive MWI notification, follow the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*}
5. **exit**
6. **voice register dn** *dn-tag*
7. **number** *number*
8. **mwi**
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP UA configuration mode for configuring the user agent.
Step 4	**mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*} [**expires** *seconds*] [**port** *port*] [**transport** {**tcp** | **udp**}] [**unsolicited**] **Example:** `Router(config-sip-ua)# mwi-server ipv4:1.5.49.200`	Specifies voice-mail server settings on a voice gateway or UA. • **ipv4**:*destinatio-address* or • **dns**: *host-name*—Keyword and argument combination to identify the mail server. • **unsolicited**—(Optional) Requires the voice-mail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.
Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 6	**voice register dn** *dn-tag* **Example:** `Router(config-register-global)# voice register dn 1`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI. • *dn-tag*—Unique sequence that identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn (voice register global)** command.
Step 7	**number** *number* **Example:** `Router(config-register-dn)# number 1234`	Defines a valid number for a directory number to be assigned to a SIP phone in Cisco Unified CME. • Enter a string of up to 16 characters that represents an E.164 telephone number. • Number string can contain alphabetic characters when the number is to be dialed only by the Cisco Unified CME router, as with an intercom number, and not from telephone keypads.

	Command or Action	Purpose
Step 8	**mwi** **Example:** `Router(config-register-dn)# mwi`	Enables a specific directory number to receive MWI notification.
Step 9	**end** **Example:** `Router(config-register-dn)# end`	Exits configuration mode and enters privileged EXEC mode.

Examples

The following example shows how to define an MWI server and specify that directory number 1, number 1234 on a SIP phone in Cisco Unified CME is to receive the MWI notification:

```
sip-ua
 mwi-server ipv4:1.5.49.200
voice register dn 1
 number 1234
 mwi
```

Where to Go Next

After integrating Cisco Unified CME and your voice mail system, you are ready to reboot or reset the phones. See the "Resetting SIP Phones in Cisco Unified CME" section on page 60.

Configuration Examples for Configuring Cisco Unified CME

The following is a configuration example for SIP phones running on Cisco Unified CME:

```
voice service voip
 allow-connections sip to sip
 sip
 registrar server expires max 600 min 60

voice class codec 1
 codec preference 1 g711ulaw

voice hunt-group 1 parallel
 final 8000
 list 2000,1000,2101
 timeout 20
 pilot 9000

voice hunt-group 2 sequential
 final 1000
 list 2000,2300
 timeout 25
 pilot 9100 secondary 9200

voice hunt-group 3 peer
 final 2300
 list 2100,2200,2101,2201
 timeout 15
 hops 3
 pilot 9300
 preference 5

voice hunt-group 4 longest-idle
 final 2000
 list 2300,2100,2201,2101,2200
 timeout 15
 hops 5
 pilot 9400 secondary 9444
 preference 5 secondary 9

voice register global
 mode cme

 external-ring bellcore-dr3

voice register dn 1
 number 2300
 mwi

voice register dn 2
 number 2200
 call-forward b2bua all 1000
 call-forward b2bua mailbox 2200
```

```
 mwi

voice register dn 3
 number 2201
 after-hour exempt

voice register dn 4
 number 2100
 call-forward b2bua busy 2000
 mwi

voice register dn 5
 number 2101
 mwi

voice register dn 76
 number 2525
 call-forward b2bua unreachable 2300
 mwi
!
voice register template 1
!
voice register template 2
 no conference enable
 voicemail 7788 timeout 5
!

voice register pool 1
 id mac 000D.ED22.EDFE
 type 7960
 number 1 dn 1
 template 1
 preference 1
 max registrations 24
 no call-waiting
 codec g711alaw
!
voice register pool 2
 id mac 000D.ED23.CBA0
 type 7960
 number 1 dn 2
 number 2 dn 2
 template 1
 preference 1
 max registrations 24
 dtmf-relay rtp-nte
 speed-dial 3 2001
 speed-dial 4 2201
!
voice register pool 3
 id mac 0030.94C3.053E
 type 7960
 number 1 dn 3
 number 3 dn 3
 template 2
 max registrations 24
!
voice register pool 5
 id mac 0012.019B.3FD8
 type ATA
 number 1 dn 5
 preference 1
 max registrations 24
 dtmf-relay rtp-nte
```

```
 codec g711alaw

voice register pool 6
 id mac 0012.019B.3E88
 type ATA
 number 1 dn 6
 number 2 dn 7
 template 2
 max registrations 24
 dtmf-relay-rtp-nte
 call-forward b2bua all 7778

voice register pool 7
 max registrations 24

voice register pool 8
 id mac 0006.D737.CC42
 type 7940
 number 1 dn 8
 template 2
 preference 1
 max registrations 24
 codec g711alaw

voice-port 1/0/0

voice-port 1/0/1

dial-peer voice 100 pots
 destination-pattern 2000
 port 1/0/0

dial-peer voice 101 pots
 destination-pattern 2010
 port 1/0/1

dial-peer voice 1001 voip
 preference 1
 destination-pattern 1...
 session protocol sipv2
 session target ipv4:10.15.6.13
 codec g711ulaw

sip-ua
 mwi-server ipv4:1.15.6.200 expires 3600 port 5060 transport udp

telephony-service
 load 7960-7940 P0S3-07-2-00
 max-ephones 24
 max-dn 96
 ip source-address 10.15.6.112 port 2000
 create cnf-files version-stamp Aug 24 2004 00:00:00
 max-conferences 8
 after-hours block pattern 1 1...
 after-hours day Mon 17:00 07:00
```

Troubleshooting Cisco Unified CME for SIP Phones

First published: October 2005

To troubleshoot SIP phones in Cisco Unified CME, perform the following task.

Finding Feature Information

Your Cisco IOS software release may not support all of the features documented in this guide. To reach links to specific feature documentation in this guide and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Unified CME for SIP Phones" section in this guide.

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Finding Support Information for Cisco Unified CME

For information about Cisco IOS software and Cisco Unified CME compatibility, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

For information about Cisco Unified CME specifications, including number of supported phones, see the appropriate *Cisco Unified CME Firmware, Platforms, Memory, and Voice Products* document at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Step 1 Enter the **show voice register statistics** command to display display statistics associated with a registration event. If *no* phones are successfully registered, perform the following steps:

a. Configure Cisco Unified CME for SIP phones.

b. Check DHCP configuration, including the default router and the TFTP location information.

c. Check that the tftp-server command is set for the required files, and use the **show voice register tftp-bind** command to check which configuration files can be accessed by SIP phones using TFTP. Use the **debug tftp events** command to monitor TFTP file access by the SIP phones during registration attempts.

d. To debug SIP registration issues, use the **debug ccsip** command.

e. To debug voice registrations, reset the phone and observe the registration attempt by entering the **debug voice register event** to display the SIP phones. Use the **voice register error** command to monitor errors during registration events by SIP phones.

f. Enter the **debug ip dhcp** command to confirm DHCP operation.

Step 2 Enter the **show voice register statistics** command to display statistics associated with a registration event. If *some* phones are successful registered, perform the following steps:

a. Enter the **show voice register all** command to display all registered phones. Check that the SIP phones show as registered.

b. Verify the IP parameter settings on the Cisco IP phone using the Settings display on the phone.

c. Reset the phone and observe the registration attempt by entering the **debug voice register event** command to display the SIP phones in Cisco UNified CME.

d. Enter the **show voice register dn** *summary* command to check the state of the Cisco IP phone lines.

e. Check the IP address of the phone, and attempt to ping the address.

Step 3 To troubleshoot basic B2BUA calls between two SIP phones in Cisco Unified CME, use the **debug ccsip message** command to display the call flow.

Step 4 To troubleshoot B2BUA CFNA, use the **debug ccsip message** and **debug voip application all** commands to display the call flow.

Step 5 To troubleshoot B2BUA call transfer, use the **debug ccsip all** command to display the call flow.

Feature Information for Cisco Unified CME for SIP Phones

Revised: June 19, 2006

This chapter lists and describes features for SIP phones in Cisco Unified CallManager Express (Cisco Unified CME).

Features that are introduced a particular Cisco IOS release are available in that and subsequent releases.

Cisco IOS software images are specific to a Cisco IOS software release, a feature set, and a platform. Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click Cancel at the login dialog box and follow the instructions that appear.

For information about the full set of Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface, glossary, and other documents—at http://www.cisco.com/en/US/products/sw/iosswrel/ps5187/prod_configuration_guide09186a008052618d.html.

Note The name of this product was changed to Cisco Unified CME in version 4.0.

.

Table 4 Feature Information for Cisco Unified CME 4.0

Feature Name	Releases	Feature Information
Dial-plan pattern expansion	12.4(4)XC	Applies dial-plan pattern expansion for call forward and call transfer when the forward or transfer-to target is an individual abbreviated SIP extension or an extension that appear on a SIP phone. See the "Configuring Dial-Plan Patterns" section on page 56 of this guide.
Voice Translation Rules	12.4(4)XC	Applies a preconfigured voice translation rule to modify the number dialed by extensions on a SIP phone. See the "Applying Voice Translation Rules" section on page 58 of this guide.

Table 5 Feature Information for Cisco CME 3.4

Feature Name	Releases	Feature Information
Bulk Registration	12.4(4)	Bulk registration of callers to a secondary external Session Initiation Protocol (SIP) Registrar. See "Configuring Bulk Registration" section on page 81.
Blocking of Unauthorized External Calls from Accessing WAN or PSTN	12.4(4)	Block unauthorized incoming SIP calls from the WAN from accessing PSTN, SIP, and H.323 trunks. See "Configuring After Hours Call Blocking" section on page 66.
Call Forward (all, busy, no answer, unreachable)	12.4(4)	Calls into a SIP device can be forwarded to other SIP or Skinny Client Control Protocol (SCCP) devices including Cisco Unity, third- party voice mail systems, or an auto-attendant (AA) or other interactive voice response (IVR) devices. SCCP devices may also be forwarded to SIP devices. See "Configuring SIP-to-SIP Phone Call Forwarding" section on page 69.
Call Waiting Hold/Resume	12.4(4)	Supports call waiting (with tone) on SIP phones. Music on hold (MOH) is not supported for call hold invoked from a SIP phone. A caller hears only silence when placed on hold by a SIP phone.
Caller ID and Name Display	12.4(4)	By default, passing a caller ID and name display to and from SIP phones is enabled. Scenarios include SIP-to-SIP, SIP-to-SCCP, SCCP-to-SIP, SIP-to-H.323 (trunk only), and H.323 (trunk only)-to-SIP is enabled. To specify the name of a person to be associated with a given directory number, see "Specifying Optional Phone-Specific Parameters for SIP Phones in Cisco Unified CME" section on page 48. To block all caller-ID displays for calls from a particular extension, see "Creating and Applying Templates to SIP Phones" section on page 88.

Table 5 Feature Information for Cisco CME 3.4 (continued)

Hunt Groups	12.4(4)	**voice hunt-group** command provides similar services as the **ephone-hunt** command does for SCCP dial peers, including sequential hunting using fixed-order, round-robin circular, and longest-idle circular selection and parallel hunting (aka call blast). This feature works with the B-ACD call queuing Tcl (Tool Command Language) scripts. See "Configuring Voice Hunt Groups" section on page 75.
Voice Mail	12.4(4)	Voice messaging systems (including Cisco Unity) connected via a SIP trunk or SIP user agent can pass a Message Waiting Indicator (MWI) that will be received and understood by a SIP phone. The standard Subscribe/Notify method is preferred over an Unsolicited Notify. The following enhancements to Message Waiting Indicator (MWI) notification are included in Cisco CME 3.4 for SIP phones: • MWI generate unsolicited notify to SIP endpoint for outcall. • Subscribe to MWI server for SIP endpoint. • Relay unsolicited notify to SIP endpoint for unsolicited notify and subscribe/notify. • Unsolicited notify interworks with MWI Relay. See the "Configuring Voice Mail Integration with Cisco Unified CME for SIP Phones" section on page 99.

Phone features vary by phone type and phone firmware releases. For additional information, see the manufacturer's documentation for your phone type and model.

.

Table 6 Phone Feature Information for Cisco CME 3.4

Feature Name	Releases	Feature Information
Call Transfer (blind, attended, semi-attended)	12.4(4)	Enables SIP phones to transfer calls to other supported SIP phones and across the SIP trunk. Similarly, Cisco CME supports the ability of SCCP phones to transfer calls to SIP phones. Calls can be transferred to a voice-mail system, such as Cisco Unity, and auto attendant or other IVR (interactive voice response) devices such as Cisco IPCC and Cisco IPCC Express. SIP-to-H.323 call transfers are not supported. Call transfer features are applied to individual SIP phones in Cisco CME by creating templates to enable common parameters. See "Creating and Applying Templates to SIP Phones" section on page 88.
Do Not Disturb	12.4(4)	The Do-not-disturb (DND) soft key enables the person receiving a call to cause the incoming call to be forwarded immediately without waiting for the no-answer time to expire or if call forward no-answer is not configured, pressing the DND soft key will mute the ringer until the call is cancelled. See "Creating and Applying Templates to SIP Phones" section on page 88 or "Configuring Do Not Disturb (DND)" section on page 93.
Intercom	12.4(4)	Creates a Cisco IP phone line connection that resembles a private line, automatic ring-down (PLAR). The auto answer causes an extension to operate in auto-dial fashion for outbound calls and auto answer with mute for inbound calls. If an extension is configured for intercom operation, it can be associated with one SIP phone only. See "Configuring Intercom Auto Answer" section on page 95.

Table 6 Phone Feature Information for Cisco CME 3.4 (continued)

Speed Dial	12.4(4)	Supports speed dialing on SIP phones. See "Configuring Speed Dial Features" section on page 96.
Three-Way Conference	12.4(4)	Supports G.711 three-way conferences among SIP phones and among combinations of SIP and SCCP phones. SIP and PSTN trunk calls may also be joined into the conference. Conference cascading is allowed so more than three parties can be included in a conference. SIP-to-H.323 calls are not supported. Music on hold (MOH) is not supported for call hold invoked from a SIP phone. A caller hears only silence when placed on hold by a SIP phone. See "Configuring Conference Call Features" section on page 90.

INDEX

D

E

F

G

H

I

K

L

M