Configuring Phones to Make Basic Calls

Last Updated: July 25, 2013

This module describes how to configure Cisco Unified IP phones in Cisco Unified Communications Manager Express (Cisco Unified CME) so that you can make and receive basic calls.

Caution The Interactive Voice Response (IVR) media prompts feature is only available on the IAD2435 when running IOS version 15.0(1)M or later.

Finding Feature Information in This Module

Your Cisco Unified CME version may not support all of the features documented in this module. For a list of the versions in which each feature is supported, see the "Feature Information for Configuring Phones to Make Basic Calls" section on page 324.

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Prerequisites for Configuring Phones to Make Basic Calls

- Cisco IOS software and Cisco Unified CME software, including phone firmware files for Cisco Unified IP phones to be connected to Cisco Unified CME, must be installed in router flash memory. See the "Installing and Upgrading Cisco Unified CME Software" section on page 61.
- For Cisco Unified IP phones that are running SIP and are connected directly to Cisco Unified CME, Cisco Unified CME 3.4 or a later version must be installed on the router. See the "Installing and Upgrading Cisco Unified CME Software" section on page 61.
- Procedures in the "Defining Network Parameters" section on page 83 and the "Configuring System-Level Parameters" section on page 119 must be completed before you start the procedures in this section.

Restrictions for Configuring Phones to Make Basic Calls

When you are configuring dial peers or ephone-dns, including park slots and conferencing extensions, on Cisco Integrated Services Router Voice Bundles, the following message may appear to warn you that free memory is not available:

%DIALPEER_DB-3-ADDPEER_MEM_THRESHOLD: Addition of dial-peers limited by available memory

To configure more dial peers or ephone-dns, increase the DRAM in the system. A moderately complex configuration may exceed the default 256 MB DRAM and require 512 MB DRAM. Note that many factors contribute to memory usage, in addition to the number of dial peers and ephone-dns configured.

Information About Configuring Phones to Make Basic Calls

To configure phones to make basic calls, you should understand the following concepts:

- Phones in Cisco Unified CME, page 191
- Directory Numbers, page 198
- Monitor Mode for Shared Lines, page 208
- Watch Mode for Phones, page 209
- PSTN FXO Trunk Lines, page 209
- Codecs for Cisco Unified CME Phones, page 210
- Analog Phones, page 212
- Secure IP Phone (IP-STE) Support, page 214
- Remote Teleworker Phones, page 217
- Busy Trigger and Channel Huntstop for SIP Phones, page 218
- Multiple Calls Per Line, page 219
- Digit Collection on SIP Phones, page 220
- Session Transport Protocol for SIP Phones, page 221
- Real-Time Transport Protocol Call Information Display Enhancement, page 221
- Ephone-Type Configuration, page 222

- Support for 7926G Wireless SCCP IP Phone, page 222
- KEM Support for Cisco Unified 8961, 9951, and 9971 SIP IP Phones, page 223
- Fast-Track Configuration Approach for Cisco Unified SIP IP Phones, page 225

Phones in Cisco Unified CME

An ephone, or "Ethernet phone," for SCCP or a voice-register pool for SIP is the software configuration for a phone in Cisco Unified CME. This phone can be either a Cisco Unified IP phone or an analog phone. Each physical phone in your system must be configured as an ephone or voice-register pool on the Cisco Unified CME router to receive support in the LAN environment. Each phone has a unique *tag*, or sequence number, to identify it during configuration.

Cisco Unified CME 8.8 and later versions support the following phones:

- Cisco Unified 3905 SIP IP Phones, page 191
- Cisco Unified 6901 and 6911 SIP IP Phones, page 192
- Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones, page 194
- Cisco Unified 8941 and 8945 SIP IP Phones, page 195
- Cisco Unified 6945, 8941, and 8945 SCCP IP Phones, page 197

Cisco Unified 3905 SIP IP Phones

Firmware 9.2(1) or a later version should be installed on the Cisco Unified 3905 SIP IP phone.

Table 13 Features Supported on the Cisco Unified 3905 SIP IP Phone

Features	Cisco Unified 3905 SIP IP Phone
After Hours	Not Supported
Authenticate Register	Supported
Auto-Answer	Supported
Barge	Not Supported
Busy-Lamp-Field Monitoring	Not Supported
Button Layout	Not Supported
Call Forward	Supported
Call Park	Not Supported
Call Transfer	Supported
cBarge	Not Supported
Conferencing	Supported
Directory Services	Not Supported
Extension Mobility	Not Supported
Group Pickup	Supported
Hold	Supported
HTTP Firmware Download	Not Supported

Table 13 Features Supported on the Cisco Unified 3905 SIP IP Phone (continued)

Features	Cisco Unified 3905 SIP IP Phone
Intercom	Not Supported
KEM	Not Supported
Live Record	Not Supported
Mobility	Not Supported
Multicast MOH	Supported
Multicast Paging	Not Supported
My Phone Apps	Not Supported
Night Service	Not Supported
Pickup	Supported
Privacy	Not Supported
Programmable Line Keys	Not Supported
Redial	Supported
Resume	Supported
Shared Lines	Supported
Speakerphone	Supported
Speed Dial	Not Supported
Unicast Paging	Not Supported
Video Telephony	Not Supported

For information on the Cisco Unified 3905 SIP IP Phone, see *Cisco Unified IP Phone 3905 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SIP).*

Cisco Unified 6901 and 6911 SIP IP Phones

Cisco Unified CME 9.0 and later versions support the Cisco Unified 6901 and 6911 SIP IP Phones.

Table 14 Features Supported on the Cisco Unified 6901 and 6911 SIP IP Phones

Features	6901	6911
After Hour	Not Supported	Not Supported
Barge	Not Supported	Not Supported
Busy-Lamp-Field Monitoring	Not Supported	Not Supported
Button Layout	Not Supported	Not Supported
Call Forward All	Supported[1]	Supported[1]
Call Park	Supported[1]	Supported[1]
Call Transfer	Supported	Supported
cBarge	Not Supported	Not Supported
Directory Service	Not Supported	Not Supported

Table 14 Features Supported on the Cisco Unified 6901 and 6911 SIP IP Phones (continued)

Features	6901	6911
Extension Mobility	Not Supported	Not Supported
Group Pickup	Supported[1]	Supported[1]
Hold	Supported	Supported
HTTP Firmware Download	Not Supported	Not Supported
Intercom	Not Supported	Not Supported
KEM	Not Supported	Not Supported
Meet-Me Conference	Not Supported	Supported[2]
Mobility	Not Supported	Not Supported
Multicast MoH	Supported	Supported
Multicast Paging	Not Supported	Supported
MyPhoneApp	Not Supported	Not Supported
Pickup	Not Supported	Supported[2]
Privacy	Not Supported	Not Supported
Programmable Line Key	Not Supported	Supported
Redial	Supported	Supported
Resume	Supported	Supported
Shared Lines	Supported	Supported
Software Ad-Hoc Conference	Supported	Supported
Speakerphone	Not Supported	Supported
Speed Dial	Not Supported	Supported
Video	Not Supported	Not Supported

1. The **fac** command must be configured in telephony-service configuration mode.
2. The **feature-button** command must be configured in voice register pool configuration mode.

Prerequisites

- Cisco IOS Release 15.2(2)T.
- Correct firmware (9.2.1 or a later version) is installed on the Cisco Unified IP phone.

Restrictions

Cisco Unified 6901 and 6911 SIP IP Phones do not have LCD screens.

For more information on the Cisco Unified 6901 and 6911 SIP IP Phones, see *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager Express Version 9.0 (SIP)*.

Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones

Cisco Unified CME 9.0 and later versions support the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones.

Table 15 Features Supported on the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones

Features	6921	6941	6945	6961
After Hour	Supported	Supported	Supported	Supported
Barge	Not Supported	Not Supported	Not Supported	Not Supported
Busy-Lamp-Field Monitoring	Supported	Supported	Supported	Supported
Button Layout	Supported	Supported	Supported	Supported
Call Forward All Softkey	Supported	Supported	Supported	Supported
Call Park	Supported	Supported	Supported	Supported
Call Transfer	Supported	Supported	Supported	Supported
cBarge	Supported	Supported	Supported	Supported
Directory Service	Supported	Supported	Supported	Supported
Extension Mobility	Supported	Supported	Supported	Supported
Group Pickup	Supported	Supported	Supported	Supported
Hold	Supported	Supported	Supported	Supported
HTTP Firmware Download	Supported	Supported	Supported	Supported
Intercom	Supported	Supported	Supported	Supported
KEM	Not Supported	Not Supported	Not Supported	Not Supported
Meet-Me Conference	Supported	Supported	Supported	Supported
Mobility	Supported	Supported	Supported	Supported
Multicast MoH	Supported	Supported	Supported	Supported
Multicast Paging	Supported	Supported	Supported	Supported
MyPhoneApp	Supported	Supported	Supported	Supported
Pickup	Supported	Supported	Supported	Supported
Privacy	Supported	Supported	Supported	Supported
Programmable Line Key	Supported	Supported	Supported	Supported
Redial	Supported	Supported	Supported	Supported
Resume	Supported	Supported	Supported	Supported
Shared Lines	Supported	Supported	Supported	Supported

Table 15 Features Supported on the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones (continued)

Features	6921	6941	6945	6961
Software Ad-Hoc Conference	Supported	Supported	Supported	Supported
Speakerphone	Supported	Supported	Supported	Supported
Speed Dial	Supported	Supported	Supported	Supported
Video	Not Supported	Not Supported	Not Supported	Not Supported

Prerequisites

- Cisco IOS Release 15.2(2)T.
- Correct firmware (9.2.1 or a later version) is installed on the Cisco Unified IP phone.

For more information on the Cisco Unified 6921, 6941, 6945, and 6961SIP IP Phones, see *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager Express Version 9.0 (SIP)*.

Cisco Unified 8941 and 8945 SIP IP Phones

Cisco Unified CME 9.0 and later versions support the Cisco Unified 8941 and 8945 SIP IP Phones.

Table 16 Features Supported on the Cisco Unified 8941 and 8945 SIP IP Phones

Features	8941	8945
After Hour	Supported	Supported
Barge	Not Supported	Not Supported
Busy-Lamp-Field Monitoring	Supported	Supported
Button Layout	Supported	Supported
Call Forward All Softkey	Supported	Supported
Call Park	Supported	Supported
Call Transfer	Supported	Supported
cBarge	Supported	Supported
Directory Service	Supported	Supported
Extension Mobility	Supported	Supported
Group Pickup	Supported	Supported
Hold	Supported	Supported
HTTP Firmware Download	Supported[1]	Supported[1]
Intercom	Supported	Supported
KEM	Not Supported	Not Supported
Meet-Me Conference	Supported	Supported

Table 16 Features Supported on the Cisco Unified 8941 and 8945 SIP IP Phones (continued)

Features	8941	8945
Mobility	Supported	Supported
Multicast MoH	Supported	Supported
Multicast Paging	Supported	Supported
MyPhoneApp	Supported	Supported
Pickup	Supported	Supported
Privacy	Supported	Supported
Programmable Line Key	Supported	Supported
Redial	Supported	Supported
Resume	Supported	Supported
Shared Lines	Supported	Supported
Software Ad-Hoc Conference	Supported	Supported
Speakerphone	Supported	Supported
Speed Dial	Supported	Supported
Video	Supported	Supported

1. For this feature, 9.2(2) or a later firmware version should be installed.

Prerequisites

- Cisco IOS Release 15.2(2)T.
- Correct firmware (9.2.1 or a later version) is installed on the Cisco Unified IP phone.

For more information on the Cisco Unified 8941 and 8945 SIP IP Phones, see *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager Express Version 9.0 (SIP)*.

Cisco Unified 6945, 8941, and 8945 SCCP IP Phones

The Cisco Unified 6945, 8941, and 8945 SCCP IP Phones are supported in Cisco Unified CME, SRST, and CME-as-SRST.

Correct firmware should be installed on the Cisco Unified IP phones:

- For Cisco Unified 6945 SCCP IP Phone, 9.1(1) or a later version.
- For Cisco Unified 8941 and 8945 SCCP IP Phones, 9.1(2) or a later version.

Table 17 Features Supported on the Cisco Unified 6945, 8941, and 8945 SCCP IP Phones

Features	6945	8941	8945
After Hours	Supported	Supported	Supported
Basic Automatic Call Distribution	Supported	Supported	Supported
Button Layout	Supported	Supported	Supported
Call Forward	Supported	Supported	Supported
Call Park	Supported	Supported	Supported
Call Transfer	Supported	Supported	Supported
Call Transfer Recall	Supported	Supported	Supported
cBarge	Not Supported	Not Supported	Not Supported
Conferencing	Supported	Supported	Supported
Directory Services	Supported	Supported	Supported
Enhanced Busy-Lamp-Field Monitoring	Supported	Supported	Supported
Extension Mobility	Supported	Supported	Supported
Forced Authorization Code	Supported	Supported	Supported
Hold	Supported	Supported	Supported
Intercom	Supported	Supported	Supported
Live Record	Supported	Supported	Supported
Multicast MOH	Supported	Supported	Supported
Multicast Paging	Supported	Supported	Supported
My Phone Apps	Supported	Supported	Supported
Night Service	Supported	Supported	Supported
Privacy	Supported	Supported	Supported
Programmable Line Keys	Supported	Supported	Supported
Resume	Supported	Supported	Supported
Secure Real-time Transport Protocol	Supported	Supported	Supported
Shared Lines	Supported	Supported	Supported
Single Number Reach	Supported	Supported	Supported
Speakerphone	Supported	Supported	Supported
Speed Dial	Supported	Supported	Supported

Table 17 Features Supported on the Cisco Unified 6945, 8941, and 8945 SCCP IP Phones (continued)

Features	6945	8941	8945
Transfer to Voicemail	Supported	Supported	Supported
Video Telephony	Supported[1]	Supported[2]	Supported[3]
Whisper Intercom	Supported	Supported	Supported

1. No built-in camera. CUVA is supported. Connection should be up between the Cisco Unified 6945 IP Phone and the Cisco Unified Video Advantage (CUVA) 2.2(1.7) or a later version.
2. With built-in camera.
3. With built-in camera.

For information on the Cisco Unified 6945 SCCP IP Phone, see *Cisco Unified IP Phone 6945 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SCCP)*.

For information on the Cisco Unified 8941 and 8945 SCCP IP Phones, see *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SCCP)*.

Directory Numbers

A directory number, also known as an ephone-dn for SCCP or a voice-register dn for SIP, is the software configuration in Cisco Unified CME that represents the line connecting a voice channel to a phone. A directory number has one or more extension or telephone numbers associated with it to allow call connections to be made. Generally, a directory number is equivalent to a phone line, but not always. There are several types of directory numbers, which have different characteristics.

Each directory number has a unique *dn-tag*, or sequence number, to identify it during configuration. Directory numbers are assigned to line buttons on phones during configuration.

One virtual voice port and one or more dial peers are automatically created for each directory number, depending on the configuration for SCCP phones, or for SIP phones, when the phone registers in Cisco Unified CME.

Because each directory number represents a virtual voice port in the router, the number of directory numbers that you create corresponds to the number of simultaneous calls that you can have. This means that if you want more than one call to the same number to be answered simultaneously, you need multiple directory numbers with the same destination number pattern.

The directory number is the basic building block of a Cisco Unified CME system. Six different types of directory numbers can be combined in different ways for different call coverage situations. Each type will help with a particular type of limitation or call-coverage need. For example, if you want to keep the number of directory numbers low and provide service to a large number of people, you might use shared directory numbers. Or if you have a limited quantity of extension numbers that you can use and you need to have a large quantity of simultaneous calls, you might create two or more directory numbers with the same number. The key is knowing how each type of directory number works and its advantages.

Not all types of directory numbers can be configured for all phones or for all protocols. In the remaining information about directory numbers, we have used SCCP in the examples presented but that does not imply exclusivity. The following sections describe the types of directory numbers in a Cisco Unified CME system:

- Single-Line, page 199
- Dual-Line, page 200
- Octo-Line, page 200
- SIP Shared-Line (Nonexclusive), page 202
- Two Directory Numbers with One Telephone Number, page 202
- Dual-Number, page 203
- Shared Line (Exclusive), page 204
- Mixed Shared Lines, page 205
- Overlaid, page 207

Single-Line

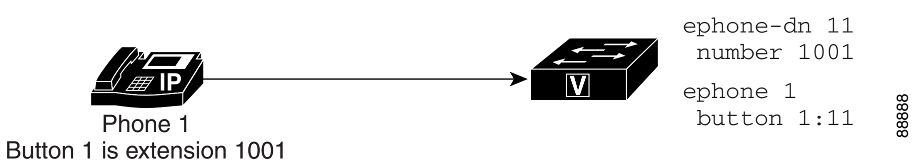
A single-line directory number has the following characteristics:

- Makes one call connection at a time using one phone line button. A single-line directory number has one telephone number associated with it.
- Should be used when phone buttons have a one-to-one correspondence to the PSTN lines that come into a Cisco Unified CME system.
- Should be used for lines that are dedicated to intercom, paging, message-waiting indicator (MWI), loopback, and music-on-hold (MOH) feed sources.
- Must have more than one single-line directory number on a phone when used with multiple-line features like call waiting, call transfer, and conferencing.
- Can be combined with dual-line directory numbers on the same phone.

Note You must make the choice to configure each directory number in your system as either dual-line or single-line when you initially create configuration entries. If you need to change from single-line to dual-line later, you must delete the configuration for the directory number, then recreate it.

Figure 7 shows a single-line directory number for an SCCP phone in Cisco Unified CME.

Figure 7 Single-Line Directory Number



Dual-Line

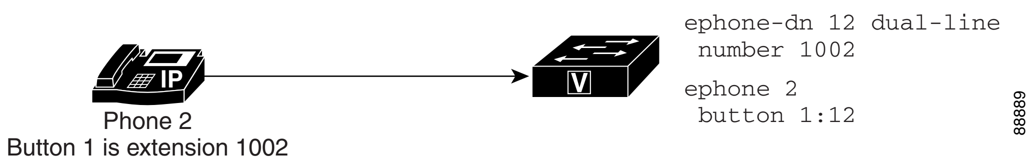
A dual-line directory number has the following characteristics:

- Has one voice port with two channels.
- Supported on IP phones that are running SCCP; not supported on IP phones that are running SIP.
- Can make two call connections at the same time using one phone line button. A dual-line directory number has two channels for separate call connections.
- Can have one number or two numbers (primary and secondary) associated with it.
- Should be used for a directory number that needs to use one line button for features like call waiting, call transfer, or conferencing.
- Cannot be used for lines that are dedicated to intercom, paging, message-waiting indicator (MWI), loopback, and music-on-hold (MOH) feed sources.
- Can be combined with single-line directory numbers on the same phone.

Note You must make the choice to configure each directory number in your system as either dual-line or single-line when you initially create configuration entries. If you need to change from single-line to dual-line later, you must delete the configuration for the directory number, then recreate it.

Figure 8 shows a dual-line directory number for an SCCP phone in Cisco Unified CME.

Figure 8 Dual-Line Directory Number



Octo-Line

An octo-line directory number supports up to eight active calls, both incoming and outgoing, on a single button of a SCCP phone. Unlike a dual-line directory number, which is shared exclusively among phones (after a call is answered, that phone owns both channels of the dual-line directory number), an octo-line directory number can split its channels among other phones that share the directory number. All phones are allowed to initiate or receive calls on the idle channels of the shared octo-line directory number.

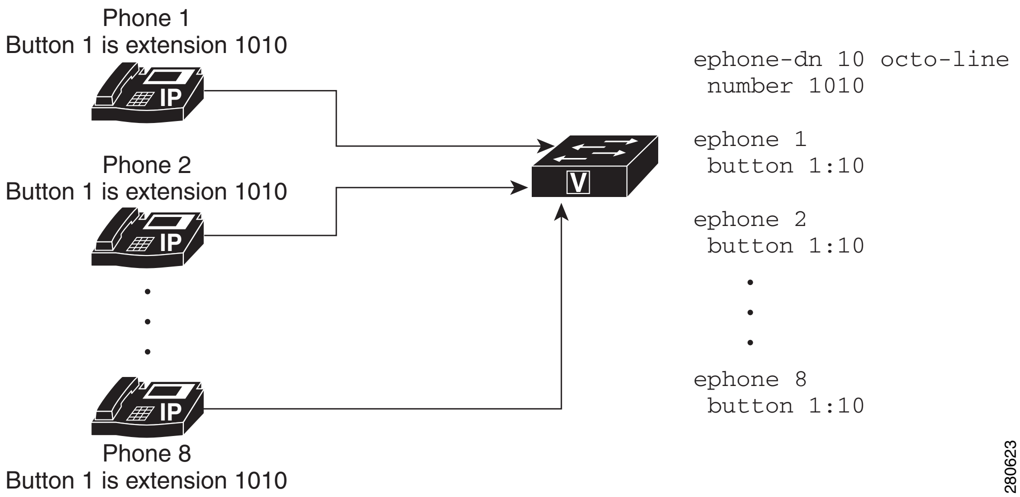
Because octo-line directory numbers do not require a different ephone-dn for each active call, one octo-line directory number can handle multiple calls. Multiple incoming calls to an octo-line directory number ring simultaneously. After a phone answers a call, the ringing stops on that phone and the call-waiting tone plays for the other incoming calls. When phones share an octo-line directory number, incoming calls ring on phones without active calls and these phones can answer any of the ringing calls. Phones with an active call hear the call-waiting tone.

After a phone answers an incoming call, the answering phone is in the connected state. Other phones that share the octo-line directory number are in the remote-in-use state.

After a connected call on an octo-line directory number is put on-hold, any phone that shares this directory number can pick up the held call. If a phone user is in the process of initiating a call transfer or creating a conference, the call is locked and other phones that share the octo-line directory number cannot steal the call.

Figure 9 shows an octo-line directory number for SCCP phones in Cisco Unified CME.

Figure 9 Octo-Line Directory Number



The Barge and Privacy features control whether other phones are allowed to view call information or join calls on the shared octo-line directory number.

Feature Comparison by Directory Number Line-Mode (SCCP Phones)

Table 18 lists some common directory number features and their support based on the type of line mode defined with the **ephone-dn** command.

Table 18 Feature Comparison by Line Mode (SCCP Phones)

Feature	Single-Line	Dual-Line	Octo-Line
Barge	—	—	Yes
Busy Trigger	—	—	Yes
Conferencing (8-party)	—	4 directory numbers	1 directory number
FXO Trunk Optimization	Yes	Yes	—
Huntstop Channel	—	Yes	Yes
Intercom	Yes	—	—
Key System (one call per button)	Yes	—	—
Maximum Calls	—	—	Yes
MWI	Yes	—	—
Overlay directory numbers (c, o, x)	Yes	Yes	—
Paging	Yes	—	—

Table 18 Feature Comparison by Line Mode (SCCP Phones) (continued)

Feature	Single-Line	Dual-Line	Octo-Line
Park	Yes	—	—
Privacy	—	—	Yes

SIP Shared-Line (Nonexclusive)

Cisco Unified CME 7.1 and later versions support SIP shared lines to allow multiple phones to share a common directory number. All phones sharing the directory number can initiate and receive calls at the same time. Calls to the shared line ring simultaneously on all phones without active calls and any of these phones can answer the incoming calls. After a phone answers a call, the ringing stops on all phones and the call-waiting tone plays for other incoming calls to the connected phone.

The phone that answers an incoming call is in the connected state. Other phones that share the directory number are in the remote-in-use state. The first user that answers the call on the shared line is connected to the caller and the remaining users see the call information and status of the shared line.

Calls on a shared line can be put on hold like calls on a nonshared line. When a call is placed on hold, other phones with the shared-line directory number receive a hold notification so all phones sharing the line are aware of the held call. Any shared-line phone user can resume the held call. If the call is placed on hold as part of a conference or call transfer operation, the call cannot be resumed by other shared-line phone users. The ID of the held call is used by other shared-line members to resume the call. Notifications are sent to all associated phones when a held call is resumed on a shared line.

Shared lines support up to 16 calls, depending on the configuration in Cisco Unified CME, which rejects any new call that exceeds the configured limit. For configuration information, see the "SIP: Creating Directory Numbers" section on page 237.

The Barge and Privacy features control whether other phones are allowed to view call information or join calls on the shared-line directory number. See the "Configuring Barge and Privacy" section on page 667.

Two Directory Numbers with One Telephone Number

Two directory numbers with one telephone or extension number have the following characteristics:

- Have the same telephone number but two separate virtual voice ports, and therefore can have two separate call connections.
- Can be dual-line (SCCP only) or single-line directory numbers.
- Can appear on the same phone on different buttons or on different phones.
- Should be used when you want the ability to make more call connections while using fewer numbers.

Figure 10 shows a phone with two buttons that have the same number, extension 1003. Each button has a different directory number (button 1 is directory number 13 and button 2 is directory number 14), so each button can make one independent call connection if the directory numbers are single-line and two call connections (for a total of four) if the directory numbers are dual-line.

Figure 11 shows two phones that each have a button with the same number. Because the buttons have different directory numbers, the calls that are connected on these buttons are independent of one another. The phone user at phone 4 can make a call on extension 1003, and the phone user on phone 5 can receive a different call on extension 1003 at the same time.

The two directory numbers-with-one-number situation is different than a shared line, which also has two buttons with one number but has only one directory number for both of them. A shared directory number will have the same call connection at all the buttons on which the shared directory number appears. If a call on a shared directory number is answered on one phone and then placed on hold, the call can be retrieved from the second phone on which the shared directory number appears. But when there are two directory numbers with one number, a call connection appears only on the phone and button at which the call is made or received. In the example in Figure 11, if the user at phone 4 makes a call on button 1 and puts it on hold, the call can be retrieved only from phone 4. For more information about shared lines, see the "Shared Line (Exclusive)" section on page 204.

The examples in Figure 10 and Figure 11 show how two directory numbers with one number are used to provide a small hunt group capability. In Figure 10, if the directory number on button 1 is busy or does not answer, an incoming call to extension 1003 rolls over to the directory number associated with button 2 because the appropriate related commands are configured. Similarly, if button 1 on phone 4 is busy, an incoming call to 1003 rolls over to button 1 on phone 5.

Figure 10 Two Directory Numbers with One Number on One Phone

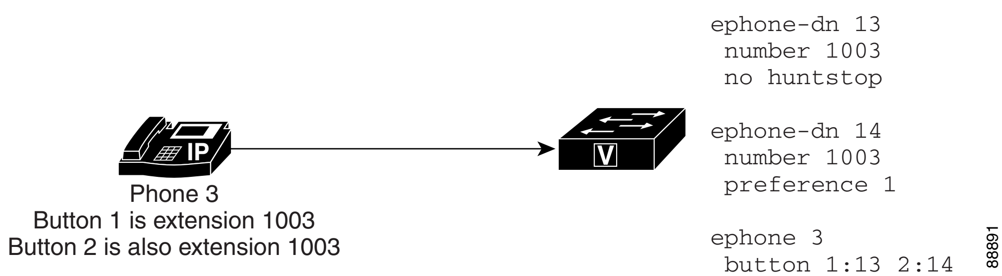
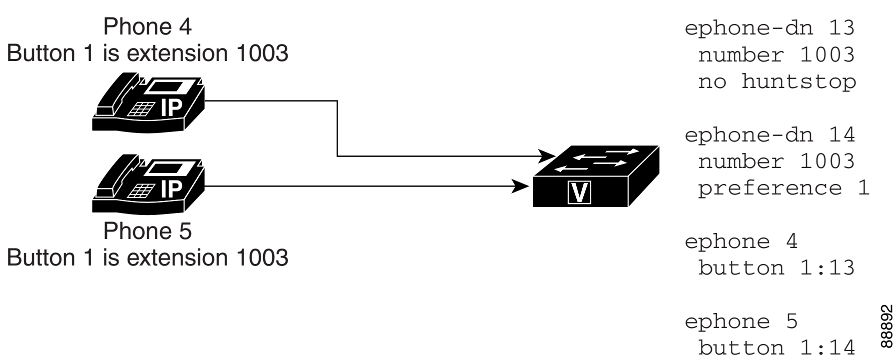


Figure 11 Two Directory Numbers with One Number on Two Phones



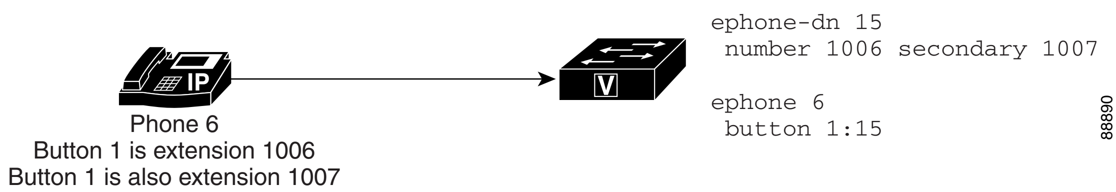
Dual-Number

A dual-number directory number has the following characteristics:

- Has two telephone numbers, a primary number and a secondary number.
- Can make one call connection if it is a single-line directory number.
- Can make two call connections at a time if it is a dual-line directory number (SCCP only).
- Should be used when you want to have two different numbers for the same button without using more than one directory number.

Figure 12 shows a directory number that has two numbers, extension 1006 and extension 1007.

Figure 12 Dual-Number Directory



Shared Line (Exclusive)

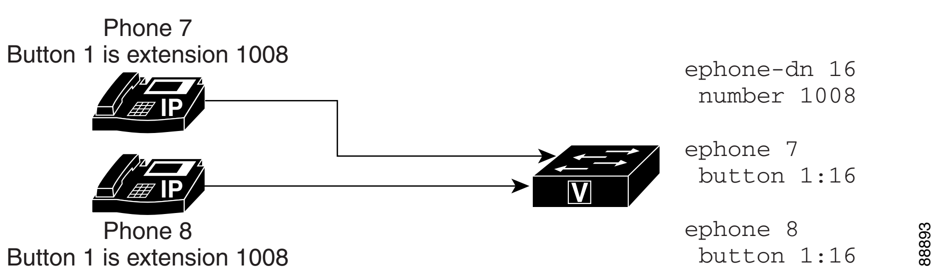
An exclusively shared directory number has the following characteristics:

- Has a line that appears on two different phones but uses the same directory number, and extension or phone number.
- Can make one call at a time and that call appears on both phones.
- Should be used when you want the capability to answer or pick up a call at more than one phone.

Because this directory number is shared exclusively among phones, if the directory number is connected to a call on one phone, that directory number is unavailable for calls on any other phone. If a call is placed on hold on one phone, it can be retrieved on the second phone. This is like having a single-line phone in your house with multiple extensions. You can answer the call from any phone on which the number appears, and you can pick it up from hold on any phone on which the number appears.

Figure 13 shows a shared directory number on phones that are running SCCP. Extension 1008 appears on both phone 7 and phone 8.

Figure 13 Shared Directory Number (Exclusive)



Mixed Shared Lines

Cisco Unified CME 9.0 and later versions support the mixed Cisco Unified SIP/SCCP shared line. This feature allows Cisco Unified SIP and SCCP IP phones to share a common directory number.

The mixed shared line supports up to 16 calls, depending on the configuration in Cisco Unified CME, which rejects any new call that exceeds the configured limit.

For configuration information, see the "SCCP: Creating Directory Numbers" section on page 227 and the "SIP: Creating Directory Numbers" section on page 237.

Incoming and Outgoing Calls

All phones sharing the common directory number can initate and receive calls at the same time. Calls to the mixed shared line ring simultaneously on all phones without active calls and any of these phones can answer the incoming calls. After a phone answers a call, the ringing stops on all phones and the call-waiting tone plays for other incoming calls to the connected phone.

The phone that answers an incoming call is in the connected state. Other phones that share the common directory number are in the remote-in-use state. The first user who answers the call on the mixed shared line is connected to the caller and the remaining users see the call information and status of the mixed shared line.

When a mixed shared-line user makes an outgoing call on the shared line, all the other shared-line users are notified of the outgoing call. When the called party answers, the caller is connected while the remaining shared-line users see the call information and the status of the call on the mixed shared line.

Hold and Resume

Calls on a mixed shared line can be put on hold like calls on a nonshared line. When a call is placed on hold, other phones with the shared-line directory number receive a hold notification so all phones sharing the line are aware of the call on hold. Any shared-line phone user can resume the call on hold. The ID of the call on hold is used by other shared-line members to resume the call. Notifications are sent to all associated phones when a call on hold is resumed on a mixed shared line. If the call is placed on hold as part of a conference or call transfer operation, the resume feature is not allowed.

Privacy on Hold

The Privacy on Hold feature prevents other phone users from viewing call information or retrieving a call put on hold by another phone sharing a common directory number. Only the caller who put the call on hold can see the status of the held call.

By default, Privacy on Hold feature is disabled for all phones on a shared line. Use the **privacy-on-hold** command in telephony-service configuration mode to enable the Privacy feature for calls that are on hold on Cisco Unified SCCP IP phones on a mixed shared line. Use the **privacy-on-hold** command in voice register global configuration mode to enable the Privacy feature for calls that are on hold on Cisco Unified SIP IP phones on a mixed shared line.

The **no privacy** and **privacy off** commands override the **privacy-on-hold** command.

Call Transfer and Forwarding

Both blind transfer and consult transfer are supported on a mixed shared line. A mixed shared line can be the one transferring the call, the one receiving the transferred call, or the call being transferred.

There are four types of call forwarding: all calls, no answer, busy, and night service. Any of these can be configured under a shared SCCP ephone-dn or a shared SIP voice register dn. However, the user must keep the call forwarding parameters for the SCCP and SIP lines synchronized with each other. A mixed shared line can be the one forwarding the call, the one receiving the forwarded call, or the call being forwarded.

For more information, see the "Configuring Call Transfer and Forwarding" section on page 763.

Call Pickup

The Call Pickup feature is supported on a mixed shared line when the **call-park system application** command is configured in telephony-service configuration mode.

A user can answer a call that:

- Originates from a shared line
- Rings on a shared line
- Originates from one shared line and rings on another shared line

For more information, see the "Call Pickup" section on page 848.

Call Park

The Call Park feature is supported on a mixed shared line when the **call-park system application** command is configured in telephony-service configuration mode.

For more information, see the "Configuring Call Park" section on page 703.

MWI

SCCP and SIP message-waiting indication (MWI) services are supported on Cisco Unity and Cisco Unity voice mails on mixed shared lines:

The following are two ways of registering a mixed shared line for an MWI service from a SIP-based MWI server with the shared-line option:

- Configure the **mwi sip** command in ephone-dn or ephone-dn-template configuration mode.
- Configure the **mwi** command in voice register dn configuration mode.

For SCCP MWI service on a mixed shared line, use the **mwi** {**off** | **on** | **on-off**} command in ephone-dn configuration mode to enable a specific Cisco Unified IP phone extension to receive MWI notification from an external voice-messaging system.

Software Conferencing

A local software conference can be created on a mixed shared line, with the mixed shared line acting as a conference creator and a conference participant.

For software conferencing on a mixed shared line, other shared-line users remain in remote-in-use state and do not see the calls on hold when the conference call is put on hold by a mixed-shared-line user acting as the conference creator.

Note Only the conference creator, who put a conference call on hold, can resume the conference call.

Dial Plans

A dial-plan pattern enables abbreviated extensions to be expanded into fully qualified E.164 numbers and builds additional dial peers for the expanded numbers it creates.

Features are effectively supported on a mixed shared line when dial-plan patterns have matching configurations in telephony-service and voice register global configuration modes using the **dialplan pattern** command.

Busy-Lamp-Field Speed-Dial Monitoring

A mixed shared line only supports directory number-based Busy-Lamp-Field (BLF) Speed-Dial monitoring and not device-based monitoring.

Restrictions

The following features are not supported on mixed Cisco Unified SIP/SCCP shared lines:

- Privacy
- Barge
- cBarge
- Single Number Reach
- Hardware Conferencing.
- Remote-resume on a local software conference call
- Video calls
- Overlay DNs on Cisco Unified SCCP IP phones
- Features in the CTI CSTA protocol suite

Overlaid

An overlaid directory number has the following characteristics:

- Is a member of an overlay set, which includes all the directory numbers that have been assigned together to a particular phone button.
- Can have the same telephone or extension number as other members of the overlay set or different numbers.
- Can be single-line or dual-line, but cannot be mixed single-line and dual-line in the same overlay set.
- Can be shared on more than one phone.

Overlaid directory numbers provide call coverage similar to shared directory numbers because the same number can appear on more than one phone. The advantage of using two directory numbers in an overlay arrangement rather than as a simple shared line is that a call to the number on one phone does not block the use of the same number on the other phone, as would happen if it were a shared directory number.

For information about configuring call coverage using overlaid ephone-dns, see the "Configuring Call Coverage Features" section on page 845.

You can overlay up to 25 lines on a single button. A typical use of overlaid directory numbers would be to create a "10x10" shared line, with 10 lines in an overlay set shared by 10 phones, resulting in the possibility of 10 simultaneous calls to the same number. For configuration information, see the "SCCP: Creating Directory Numbers for a Simple Key System" section on page 258

Monitor Mode for Shared Lines

In Cisco CME 3.0 and later versions, monitor mode for shared lines provides a visible line status indicating whether the line is in-use or not. A monitor-line lamp is off or unlit only when its line is in the idle call state. The idle state occurs before a call is made and after a call is completed. For all other call states, the monitor line lamp is lit. A receptionist who monitors the line can see that it is in use and can decide not to send additional calls to that extension, assuming that other transfer and forwarding options are available, or to report the information to the caller; for example, "Sorry, that extension is busy, can I take a message?"

In Cisco CME 3.2 and later versions, consultative transfers can occur during Direct Station Select (DSS) for transferring calls to idle monitored lines. The receptionist who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line. For information about consultative transfer with DSS, see the "Configuring Call Transfer and Forwarding" section on page 763.

In Cisco Unified CME 4.0(1) and later versions, the line button for a monitored line can be used as a DSS for a call transfer when the monitored line is idle or in-use*,* provided that the call transfer can succeed; for example, when the monitored line is configured for Call Forward Busy or Call Forward No Answer.

Note Typically, Cisco Unified CME does not attempt a transfer that causes the caller (transferee) to hear a busy tone. However, the system does not check the state of subsequent target numbers in the call-forward path when the transferred call is transferred more than once. Multiple transfers can occur because a call-forward-busy target is also busy and configured for Call Forward Busy.

In Cisco Unified CME 4.3 and later versions, a receptionist can use the Transfer to Voicemail feature to transfer a caller directly to a voice-mail extension for a monitored line. For configuration information, see the "SCCP: Enabling Transfer to Voice Mail" section on page 550.

For configuration information for monitor mode, see the "SCCP: Assigning Directory Numbers to Phones" section on page 233.

Monitor mode is intended for use only in the context of shared lines so that a receptionist can visually monitor the in-use status of several users' phone extensions; for example, for Busy Lamp Field (BLF) notification. To monitor all lines on an individual phone so that a receptionist can visually monitor the in-use status of that phone, see the "Watch Mode for Phones" section on page 209.

For BLF monitoring of speed-dial buttons and directory call-lists, see the "Configuring Presence Service" section on page 1277.

Watch Mode for Phones

In Cisco Unified CME 4.1 and later versions, a line button that is configured for watch mode on one phone provides BLF notification for all lines on another phone (watched phone) for which watched directory number is the primary line. Watch mode allows a phone user, such as a receptionist, to visually monitor the in-use status of an individual phone. A user can use the line button that has been set in watch mode as a speed-dial to call the first extension of the watched phone. The watching phone button displays a red light when the watched phone is unregistered in a DND state or in an offhook state. Pressing the button when it is not displaying a red light will dial the number in the same manner it would for a monitor button or the speed-dial button. Incoming calls on a line button that is in watch mode do not ring and do not display caller ID or call-waiting caller ID.

The line button for a watched phone can also be used as a DSS for a call transfer when the watched phone is idle. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the watched directory number, causing the call to be transferred to the phone number associated with the watched directory number.

For configuration information, see the "SCCP: Assigning Directory Numbers to Phones" section on page 233.

If the watched directory number is a shared line and the shared line is not idle on any phone with which it is associated, then in the context of watch mode, the status of the line button indicates that the watched phone is in use.

For best results when monitoring the status of an individual phone based on a watched directory number, the directory number configured for watch mode should not be a shared line. To monitor a shared line so that a receptionist can visually monitor the in-use status of several users' phone extensions, see the "Monitor Mode for Shared Lines" section on page 208.

For BLF monitoring of speed-dial buttons and directory call-lists, see the "Configuring Presence Service" section on page 1277.

PSTN FXO Trunk Lines

In Cisco CME 3.2 and later versions, IP phones running SCCP can be configured to have buttons for dedicated PSTN FXO trunk lines, also known as FXO lines. FXO lines may be used by companies whose employees require private PSTN numbers. For example, a salesperson may need a special number that customers can call without having to go through a main number. When a call comes in to the direct number, the salesperson knows that the caller is a customer. In the salesperson's absence, the customer can leave a voice mail. FXO lines can use PSTN service provider voice mail: when the line button is pressed, the line is seized, allowing the user to hear the stutter dial tone provided by the PSTN to indicate that voice messages are available.

Because FXO lines behave as private lines, users do not have to dial a prefix, such as 9 or 8, to reach an outside line. To reach phone users within the company, FXO-line users must dial numbers that use the company's PSTN number. For calls to non-PSTN destinations, such as local IP phones, a second directory number must be provisioned.

Calls placed to or received on an FXO line have restricted Cisco Unified CME services and cannot be transferred by Cisco Unified CME. However, phone users are able to access hookflash-controlled PSTN services using the Flash soft key.

In Cisco Unified CME 4.0(1), the following FXO trunk enhancements were introduced to improve the keyswitch emulation behavior of PSTN lines on phones running SCCP in a Cisco Unified CME system:

- FXO port monitoring—Allows the line button on IP phones to reliably show the status of an FXO port when the port is in use. The status indicator, either a lamp or an icon, depending on the phone model, accurately displays the status of the FXO port during the duration of the call, even after the call is forwarded or transferred. The same FXO port can be monitored by multiple phones using multiple trunk ephone-dns.
- Transfer recall—If a transfer-to phone does not answer after a specified timeout, the call is returned to the phone that initiated the transfer and it resumes ringing on the FXO line button. The directory number must be dual-lined.
- Transfer-to button optimization—When an FXO call is transferred to a private extension button on another phone, and that phone has a shared line button for the FXO port, after the transfer is committed and the call is answered, the connected call displays on the FXO line button of the transfer-to phone. This frees up the private extension line on the transfer-to phone. The directory number *n* must be dual-line.
- Dual-line ephone-dns— Directory numbers for FXO lines can now be configured for dual-line to support the FXO monitoring, transfer recall, and transfer-to button optimization features.

For configuration information, see the "SCCP: Configuring Trunk Lines for a Key System" section on page 261.

Codecs for Cisco Unified CME Phones

In Cisco CME 3.4, support for connecting and provisioning SIP phones was added. The default codec of the POTS dial peer for an SCCP phone is G.711 and the default codec of a VoIP dial peer for a SIP phone is G.729. If neither the SCCP phone nor the SIP phone in Cisco Unified CME is specifically configured to change the codec, calls between the two phones on the same router will produce a busy signal caused by the mismatched default codecs. To avoid codec mismatch, specify the codec for individual IP phones in Cisco Unified CME. Modify the configuration for either SIP or SCCP phones to ensure that the codec for all phones match. Do not modify the configuration for both SIP and SCCP phones. For configuration information, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.

In Cisco Unified CME 4.3, support for G.722-64K and the Internet Low Bit Rate Codec (iLBC) was added. This enables Cisco Unified CME to support the same codecs that are used in newer Cisco Unified IP phones, mobile wireless networks, and internet telephony without transcoding. This feature provides support for the following:

- iLBC and G.722-capable SIP and SCCP IP phones in Cisco Unified CME.
- iLBC-capable SCCP analog endpoints and remote phones in Cisco Unified CME.
- Conferencing support for G.722 and ILBC.
- Supplementary services, such as transfer, call forward, MOH, support for G.722 and iLBC, including any supplementary services that require transcoding between G.722 and any other codec.
- Transcoding for G.722 and iLBC, including G.722 to G.711 and G.722 to any other codec.

With the introduction of G.722 and iLBC codecs, there can be a disparity between codec capabilities of different phones and different firmware versions on same phone type. For example, when a H.323 call is established, the codec is negotiated based on the dial-peer codec and the assumption is that the codecs supported on H.323 side are supported by the phones. This assumption is not valid after G.722 and ILBC codec are introduced in your network. If the phones do not support the codecs on the H.323 side, a transcoder is required. To avoid transcoding in this situation, configure incoming dial-peers so that

G.722 and iLBC codecs are not used for calls to phones that are not capable of supporting these codecs. Instead, configure these phones for G.729 or G.711. Also, when configuring shared directory numbers, ensure that phones with the same codec capabilities are connected to the shared directory number.

G.722-64K

Traditional PSTN telephony codecs, including G.711 and G.729, are classified as narrowband codecs because they encode audio signals in a narrow audio bandwidth, giving telephone calls a characteristic "tinny" sound. Wideband codecs, such as G.722, provide a superior voice experience because wideband frequency response is 200 Hz to 7 kHz compared to narrowband frequency response of 300 Hz to 3.4 kHz. At 64 kbps, the G.722 codec offers conferencing performance and good music quality.

A wideband handset for certain Cisco Unified IP phones, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G-GE, 7942G, 7945G, 7961G-GE, 7962G, 7965G, and 7975G, take advantage of the higher voice quality provided by wideband codecs to enhance end-user experience with high-fidelity wideband audio. When users use a headset that supports wideband, they experience improved audio sensitivity when the wideband setting on their phones is enabled. You can configure phone-user access to the wideband headset setting on IP phones by setting the appropriate VendorConfig parameters in the phone's configuration file. For configuration information, see the "Modifying Cisco Unified IP Phone Options" section on page 1457.

If the system is not configured for a wideband codec, phone users may not detect any additional audio sensitivity, even when they are using a wideband headset.

You can configure the G.722-64K codec at a system-level for all calls through Cisco Unified CME. For configuration information, see the "Modifying the Global Codec" section on page 254. To configure individual phones and avoid codec mismatch for calls between local phones, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.

iLBC codec

Internet Low Bit Rate Codec (iLBC) enables graceful speech quality degradation in a network where frames get lost. Consider iLBC suitable for real-time communications, such as telephony and video conferencing, streaming audio, archival, and messaging. This codec is widely used by internet telephony softphones.The SIP, SCCP, and MGCP call protocols support use of the iLBC as an audio codec. iLBC provides better voice quality than G.729 but less than G.711. Supporting codecs that have standardized use in other networks, such as iLBC, enables end-to-end IP calls without the need for transcoding.

To configure individual SIP or SCCP phones, including analog endpoints in Cisco Unified CME, and avoid codec mismatch for calls between local phones, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.

Analog Phones

Cisco Unified CME supports analog phones and fax machines using Cisco Analog Telephone Adaptors (ATAs) or FXS ports in SCCP, H.323 mode, and fax pass-through mode. The FXS ports used for analog phones or fax can be on a Cisco Unified CME router, Cisco VG224 voice gateway, or integrated services router (ISR).

This section provides information on the following topics:

- Cisco ATAs in SCCP Mode, page 212
- FXS Ports in SCCP Mode, page 212
- FXS Ports in H.323 Mode, page 212
- Fax Support, page 213
- Cisco VG202, VG204, and VG224 Autoconfiguration, page 214

Cisco ATAs in SCCP Mode

You can configure the Cisco ATA 186 or Cisco ATA 188 to cost-effectively support analog phones using SCCP in Cisco IOS Release 12.2(11)T and later versions. Each Cisco ATA enables two analog phones to function as IP phones. For configuration information, see the "Configuring Cisco ATA Support" section on page 272.

FXS Ports in SCCP Mode

FXS ports on Cisco VG224 Voice Gateways and Cisco 2800 Series and Cisco 3800 Series ISRs can be configured for SCCP supplementary features. For information about using SCCP supplementary features on analog FXS ports on a Cisco IOS gateway under the control of a Cisco Unified CME router, see *Supplementary Services Features for FXS Ports on Cisco IOS Voice Gateways Configuration Guide*.

FXS Ports in H.323 Mode

FXS ports on platforms that cannot enable SCCP supplementary features can use H.323 mode to support call waiting, caller ID, hookflash transfer, modem pass-through, fax (T.38, Cisco fax relay, and pass-through), and PLAR. These features are provisioned as Cisco IOS voice features and not as Cisco Unified CME features.

Note When using Cisco Unified CME, you can configure FXS ports in H.323 mode for call waiting or hookflash transfer, but not both at the same time.

See the following documents for details on configuring features for FXS ports in H.323 mode:

- "Configuring Analog Voice Ports" section in *Voice Ports Configuration Guide*
- "Caller ID" document in *Cisco IOS Voice Configuration Library*
- *Cisco IOS Fax, Modem, and Text Support over IP Application Guide*

Fax Support

Cisco Unified CME 4.0 introduced the use of G.711 fax pass-through for SCCP on the Cisco VG224 voice gateway and Cisco ATA. In Cisco Unified CME 4.0(3) and later versions, fax relay using the Cisco-proprietary fax protocol is the only supported fax option for SCCP-controlled FXS ports on the Cisco VG224 and integrated service routers. For more information on fax relay, see the "Configuring Fax Relay" section on page 1161.

Cisco ATA-187

Cisco Unified CME 9.0 and later versions provide voice and fax support on Cisco ATA-187.

Cisco ATA-187 is a SIP-based analog telephone adaptor that turns traditional telephone devices into IP devices. Cisco ATA-187 can connect with a regular analog FXS phone or fax machine on one end, while the other end is an IP side that uses SIP for signaling and registers to Cisco Unified CME as a Cisco Unfiied SIP IP phone.

Cisco ATA-187 functions as a Cisco Unified SIP IP phone that supports T.38 fax relay and fax pass-through, enabling the real-time transmission of fax over IP networks. The fax rate is from 7.2 to 14.4 kbps.

For information on how to configure voice and fax support on Cisco ATA-187, see the "Configuring Voice and T.38 Fax Relay on Cisco ATA-187" section on page 277.

Table 19 Features Supported on Cisco ATA-187

Features	ATA-187
Ad-Hoc Conference (Hardware DSP)	Not Supported
Ad-Hoc Conference (Three-Way)	Supported[1]
Barge	Not Supported
Call Forward All	Supported
Call Transfer	Supported
Call Waiting	Supported
cBarge	Not Supported
Hold	Supported
Meet-Me Conference	Supported
Pickup	Supported
Redial	Supported
Resume	Supported
Shared Lines	Supported
Speed Dial	Supported
Voice Mail	Supported

1. For this feature, 9.2(3) or a later firmware version should be installed.

For more information on Cisco ATA-187, see *Cisco ATA 187 Analog Telephone Adaptor Administration Guide for SIP*.

Cisco VG202, VG204, and VG224 Autoconfiguration

The Autoconfiguration feature in Cisco Unified CME 7.1 and later versions allows you to automatically configure the Cisco VG202, VG204, and VG224 Analog Phone Gateway. You can configure basic voice gateway information in Cisco Unified CME, which then generates XML configuration files for the gateway and saves the files to either the default location in system:/its/ or to a location you define in system memory, flash memory, or an external TFTP server. When the voice gateway powers up, it downloads the configuration files from Cisco Unified CME and based on the information in the files, the voice gateway provisions its analog voice ports and creates the corresponding dial peers.

Using this Autoconfiguration feature with the existing Auto Assign feature allows you to quickly set up analog phones to make basic calls. After the voice gateway is properly configured and it downloads its XML configuration files from Cisco Unified CME, the SCCP telephony control (STC) application registers each configured voice port to Cisco Unified CME.

If you enable the Auto Assign feature, the gateway automatically assigns the next available directory number from the pool set by the **auto assign** command, binds that number to the requesting voice port, and creates an ephone entry associated with the voice port. The MAC address for the ephone entry is calculated based on the MAC address of the gateway and the port number. You can manually assign a directory number to each of the voice ports by creating the ephone-dn and corresponding ephone entry.

You can initiate a reset or restart of the analog endpoints from Cisco Unified CME, which triggers the autoconfiguration process. The voice gateway downloads its configuration files from Cisco Unified CME and applies the new changes.

For configuration information, see the "SCCP: Enabling Auto-Configuration for Cisco VG202, VG204, and VG224" section on page 281.

Secure IP Phone (IP-STE) Support

Cisco Unified CME 8.0 adds support for a new secure endpoint, Internet Protocol - Secure Telephone Equipment (IP-STE). IP-STE is a standalone, V.150.1 capable device which functions like a 7960 phone with secure communication capability. IP-STE has native state signaling events (SSE / SPRT) support and supports SCCP protocol. IP-STE uses the device ID 30035 when registering to a SCCP server. However, only V.150.1 modem relay is implemented in an IP-STE stack and V150.1 modem passthrough is not supported. Therefore, the response to capability query from Cisco Unified CME only includes media_payload_XV150_MR_711U and media_payload_xv150_MR_729A.

For configuration information, see the "SCCP: Configuring Secure IP Phone (IP-STE)" section on page 292.

The following support is added for IP-STE endpoints:

- The IP-STE endpoint allows secure communication between gateway-connected legacy analog STE/STU devices and IP STE devices using existing STE devices in voice networks.
- Secure voice and secure data modes from STE/STU devices connected to Cisco IOS gateway foreign exchange station (FXS) and BRI ports to an IP-STE.
- Support for the state signaling events (SSE) protocol, allowing for modem signaling end-to-end and VoIP to modem over IP (MoIP) transition and operation.
- Interoperation between line-side and trunk-side gateways and Cisco Unified CME to determine codec support and V.150.1 negotiation. You can configure gateway-attached devices to support either modem relay, modem pass-through, both modem transport methods, or neither method.

This section provides information on the following topics:

- Secure Communications Between STU, STE, and IP-STE, page 215
- SCCP Media Control for Secure Mode, page 215
- Secure Communication Between STE, STU, and IP-STE Across SIP Trunk, page 216

Secure Communications Between STU, STE, and IP-STE

Secure Telephone Equipment (STE) and Secure Telephone Units (STUs) encrypt voice and data streams with government proprietary algorithms (Type-1 encryption). To provide support for the legacy STEs and STUs and next generation IP Secure Telephone Equipment (IP-STE), voice gateways must be able to support voice and data in secure mode within the IP network and be able to pass calls within and also to and from government voice networks.

In earlier versions of Cisco Unified CME, Cisco IOS gateways supported secure voice and data communication between legacy STE and STU devices using modem pass-through method. Cisco Unified CME 8.0 and later versions control the secure endpoints by implementing a subset of v.150.1 modem relay protocol and ensures secure communications between IP-STE endpoints and STE/STU endpoints. This allows Cisco Unified CME SCCP controlled secure endpoints to communicate with the IP-STE or legacy endpoints in secure mode.

SCCP Media Control for Secure Mode

IP-STE endpoints use the V.150.1 modem relay transport method using Future Narrow Band Digital Terminal (FNBDT) signaling over a V.32 or V.34 data pump for secure communication with other legacy STE endpoints. However, IP-STE endpoints cannot communicate with STU endpoints because STU endpoints use the modem pass-through method using a proprietary data pump and do not support the FNBDT signalling.

Secure communication between IP-STE endpoints and legacy STE endpoints support the following encryption-capable endpoints:

- STE—Specialized encryption-capable analog or BRI phones that can communicate over V.150.1 modem relay or over modem pass-through, also known as Voice Band Data (VBD).
- IP-STE—Specialized encryption-capable IP phones that communicate only over V.150.1 modem relay.
- STU—Specialized encryption-capable analog phones that operate only over NSE-based modem pass-through connections.

Table 20 lists call scenarios between devices along with modem transport methods that the IP-STE endpoints use to communicate with STE endpoints.

Table 20 Supported Secure Call Scenarios and Modem Transport Methods

Device Type	STU	STE	IP-STE
STU	Pass-through	Pass-through	None
STE	Pass-through	Pass-through	Relay
IP-STE	None	Relay	Relay

Secure Communication Between STE, STU, and IP-STE Across SIP Trunk

The Secure Device Provisioning (SDP) for SIP end-to end negotiation includes four proprietary media types for secure communication between Cisco Unified CME and SIP trunk. These proprietary VBD or Modem Relay (MR) media types can be encoded into media attributes of SDP media lines. VBD capabilities are signaled using the SDP extension mechanism and Cisco proprietary nomenclature. MR capabilities are signaled through V.150.1. The following example shows VBD capabilities. The SDP syntax are based on RFC 2327 and V.150.1 Appendix E.

```
a=rtpmap:100 X-NSE/8000
a=rtpmap:118 v150fw/8000
a=sqn:0
a=cdsc:1 audio RTP/AVP 118 0 18
a=cdsc: 4 audio udsprt 120
a=cpar: a=sprtmap: 120 v150mr/8000
```

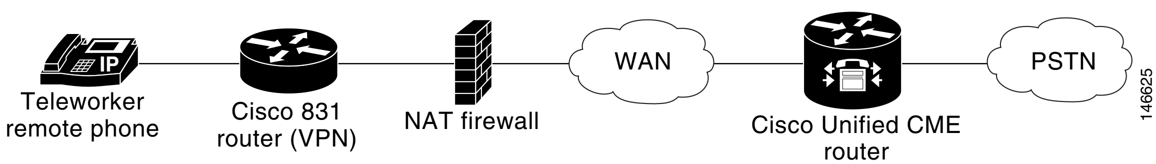
Remote Teleworker Phones

IP phones or a Cisco IP Communicator can be connected to a Cisco Unified CME system over a WAN to support teleworkers who have offices that are remote from the Cisco Unified CME router. The maximum number of remote phones that can be supported is determined by the available bandwidth.

IP addressing is a determining factor in the most critical aspect of remote teleworker phone design. The following two scenarios represent the most common designs, the second one is the most common for small and medium businesses:

- Remote site IP phones and the hub Cisco Unified CME router use globally routable IP addresses.
- Remote site IP phones use NAT with unroutable private IP addresses and the hub Cisco Unified CME router uses a globally routable address (see Figure 14). This scenario results in one-way audio unless you use one of the following workarounds:
  - Configure static NAT mapping on the remote site router (for example, a Cisco 831 Ethernet Broadband Router) to convert between a private address and a globally routable address. This solution uses fewer Cisco Unified CME resources, but voice is unencryped across the WAN.
  - Configure an IPsec VPN tunnel between the remote site router (For example, a Cisco 831 Ethernet Broadband Router) and the Cisco Unified CME router. This solution requires Advanced IP Services or higher image on the Cisco Unified CME router if this router is used to terminate the VPN tunnel. Voice will be encrypted across the WAN. This method will also work with the Cisco VPN client on a PC to support a Cisco IP Communicator.

Figure 14 Remote Site IP Phones Using NAT



Media Termination Point for Remote Phones

Media termination point (MTP) configuration is used to ensure that Real-Time Transport Protocol (RTP) media packets from remote phones always transit through the Cisco Unified CME router. Without the MTP feature, a phone that is connected in a call with another phone in the same Cisco Unified CME system sends its media packets directly to the other phone, without the packets going through the Cisco Unified CME router. MTP forces the packets to be sourced from the Cisco Unified CME router.

When this configuration is used to instruct a phone to always send its media packets to the Cisco Unified CME router, the router acts as an MTP or proxy and forwards the packets to the destination phone. If a firewall is present, it can be configured to pass the RTP packets because the router uses a specified UDP port for media packets. In this way, RTP packets from remote IP phones can be delivered to IP phones on the same system though they must pass through a firewall.

You must use the **mtp** command to explicitly enable MTP for each remote phone that sends media packets to Cisco Unified CME.

One factor to consider is whether you are using multicast music on hold (MOH) in your system. Multicast packets generally cannot be forwarded to phones that are reached over a WAN. The multicast MOH feature checks to see if MTP is enabled for a phone and if it is, MOH is not sent to that phone. If you have a WAN configuration that can forward multicast packets and you can allow RTP packets through your firewall, you can decide not to use MTP.

For configuration information, see the "SCCP: Enabling a Remote Phone" section on page 287.

G.729r8 Codec on Remote Phones

You can select the G.729r8 codec on a remote IP phone to help save network bandwidth. The default codec is G.711 mu-law. If you use the **codec g729r8** command without the **dspfarm-assist** keyword, the use of the G.729 codec is preserved only for calls between two phones on the Cisco Unified CME router (such as between an IP phone and another IP phone or between an IP phone and an FXS analog phone). The **codec g729r8** command has no affect on a call directed through a VoIP dial peer unless the **dspfarm-assist** keyword is also used.

For configuration information, see the "SCCP: Enabling a Remote Phone" section on page 287.

For information about transcoding behavior when using the G.729r8 codec, see the "Transcoding When a Remote Phone Uses G.729r8" section on page 453.

Busy Trigger and Channel Huntstop for SIP Phones

Cisco Unified CME 7.1 introduced busy trigger and huntstop channel support for SIP phones, such as the Cisco Unified IP Phone 7941G, 7941GE, 7942G, 7945G, 7961G, 7961GE, 7962G, 7965G, 7970G, 7971GE, 7975G, and 7985. For these SIP phones, the number of channels supported is limited by the amount of memory on the phone. To prevent incoming calls from overloading the phone, you can configure a busy trigger and a channel huntstop for the directory numbers on the phone.

The Channel Huntstop feature limits the number of channels available for incoming calls to a directory number. If the number of incoming calls reaches the configured limit, Cisco Unified CME does not present the next incoming call to the directory number. This reserves the remaining channels for outgoing calls or for features, such as call transfer and conferencing.

The Busy Trigger feature limits the calls to a directory number by triggering a busy response. After the number of active calls, both incoming and outgoing, reaches the configured limit, Cisco Unified CME forwards the next incoming call to the Call Forward Busy destination or rejects the call with a busy tone if Call Forward Busy is not configured.

The busy-trigger limit applies to all directory numbers on a phone. If a directory number is shared among multiple SIP phones, Cisco Unified CME presents incoming calls to those phones that have not reached their busy-trigger limit. Cisco Unified CME initiates the busy trigger for an incoming call only if all the phones sharing the directory number exceed their limit.

For configuration information, see the "SIP: Creating Directory Numbers" section on page 237 and the "SIP: Assigning Directory Numbers to Phones" section on page 240.

Multiple Calls Per Line

Cisco Unified CME 9.0 provides support for the Multiple Calls Per Line (MCPL) feature on Cisco Unified 6921, 6941, 6945, and 6961 SIP IP phones and Cisco Unified 8941 and 8945 SCCP and SIP IP phones.

Before Cisco Unified CME 9.0, the maximum number of calls supported for every directory number (DN) on Cisco Unified 8941 and 8945 SCCP IP phones was restricted to two.

With Cisco Unified CME 9.0, the MCPL feature overcomes the limitation on the maximum number of calls per line.

In Cisco Unified CME 9.0, the MCPL feature is not supported on Cisco Unified 6921, 6941, 6945, and 6961 SCCP IP phones.

Cisco Unified 8941 and 8945 SCCP IP Phones

Before Cisco Unified CME 9.0, Cisco Unified 8941 and 8945 SCCP IP phones only supported two incoming calls per line and a third channel was reserved for call transfers or conference calls. These phones were also hardcoded with **ephone-dn octo-line**, **huntstop-channel** 2, **max-calls -per-button** 3, and **busy-trigger-per-button** 2 .

In Cisco Unified CME 9.0, you can configure the **ephone-dn dn-tag** [**dual-line** | **octo-line**] in global configuration mode and the **max-calls-per-button** and **busy-trigger-per-button** commands in ephone or ephone-template configuration mode for Cisco Unified 8941 and 8945 SCCP IP phones to configure a DN and enable the number of calls per DN, set the maximum number of calls allowed on an octo-line DN, and set the maximum number of calls allowed on an octo-line DN before activating a busy tone.

For configuration information, see the "SCCP: Configuring the Maximum Number of Calls" section on page 300.

Cisco Unified 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP Phones

In Cisco Unified CME 9.0, the default values for the **busy-trigger-per-button** command is 1 for the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP phones and 2 for the Cisco Unified 8941 and 8945 SIP IP phones.

You can configure the maximum number of calls before a phone receives a busy tone. For example, if you configure **busy-trigger-per-button** 2 in voice register pool configuration mode for a Cisco Unified 6921, 6941, 6945, or 6961 SIP IP phone, the third incoming call to the phone receives a busy tone.

For information on the Busy Trigger feature on Cisco Unified SIP IP phones, see the "Busy Trigger and Channel Huntstop for SIP Phones" section on page 218.

For configuration information, see the "SIP: Configuring the Busy Trigger Limit" section on page 303.

Digit Collection on SIP Phones

Digit strings dialed by phone users must be collected and matched against predefined patterns to place calls to the destination corresponding to the user's input. Before Cisco Unified CME 4.1, SIP phone users had to press the DIAL soft key or # key or wait for the interdigit-timeout to trigger call processing. In Cisco Unified CME 4.1 and later versions, two methods of collecting and matching digits are supported for SIP phones, depending on the model of phone:

- KPML Digit Collection, page 220
- SIP Dial Plans, page 220

KPML Digit Collection

Key Press Markup Language (KPML) uses SIP SUBSCRIBE and NOTIFY methods to report user input digit by digit. Each digit dialed by the phone user generates its own signaling message to Cisco Unified CME, which performs pattern recognition by matching a destination pattern to a dial peer as it collects the dialed digits. This process of relaying each digit immediately is similar to the process used by SCCP phones. It eliminates the need for the user to press the Dial soft key or wait for the interdigit timeout before the digits are sent to Cisco Unified CME for processing.

KPML is supported on Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE. For configuration information, see the "SIP: Enabling KPML" section on page 248.

SIP Dial Plans

A dial plan is a set of dial patterns that SIP phones use to determine when digit collection is complete after a user goes off-hook and dials a destination number. Dial plans allow SIP phones to perform local digit collection and recognize dial patterns as user input is collected. After a pattern is recognized, the SIP phone sends an INVITE message to Cisco Unified CME to initiate the call to the number matching the user's input. All of the digits entered by the user are presented as a block to Cisco Unified CME for processing. Because digit collection is done by the phone, dial plans reduce signaling messages overhead compared to KPML digit collection.

SIP dial plans eliminate the need for a user to press the Dial soft key or # key or to wait for the interdigit timeout to trigger an outgoing INVITE. You configure a SIP dial plan and associate the dial plan with a SIP phone. The dial plan is downloaded to the phone in the configuration file.

You can configure SIP dial plans and associate them with the following SIP phones:

- Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE—These phones use dial plans and support KPML. If both a dial plan and KPML are enabled, the dial plan has priority.

  If a matching dial plan is not found and KPML is disabled, the user must wait for the interdigit timeout before the SIP NOTIFY message is sent to Cisco Unified CME. Unlike other SIP phones, these phones do not have a Dial soft key to indicate the end of dialing, except when on-hook dialing is used. In this case, the user can press the Dial soft key at any time to send all the dialed digits to Cisco Unified CME.
- Cisco Unified IP Phones 7905, 7912, 7940, and 7960—These phones use dial plans and do not support KPML. If you do not configure a SIP dial plan for these phones, or if the dialed digits do not match a dial plan, the user must press the Dial soft key or wait for the interdigit timeout before digits are sent to Cisco Unified CME.

When you reset a phone, the phone requests its configuration files from the TFTP server, which builds the appropriate configuration files depending on the type of phone.

- Cisco Unified IP Phones 7905 and 7912—The dial plan is a field in their configuration files.
- Cisco Unified IP Phones 7911G, 7940, 7941G, 7941GE, 7960, 7961G, 7961GE, 7970G, and 7971GE—The dial plan is a separate XML file that is pointed to from the normal configuration file.

For configuration information for Cisco Unified CME, see the "SIP: Configuring Dial Plans" section on page 243.

Session Transport Protocol for SIP Phones

In Cisco Unified CME 4.1 and later versions, you can select TCP as the transport protocol for connecting supported SIP phones to Cisco Unified CME. Previously only UDP was supported. TCP is selected for individual SIP phones by using the **session-transport** command in voice register pool or voice register template configuration mode. For configuration information, see the "SIP: Selecting Session-Transport Protocol for a Phone" section on page 250.

Real-Time Transport Protocol Call Information Display Enhancement

Before Cisco Unified CME 8.8, active RTP call information on ephone call legs were determined only by parsing the **show ephone registered** or **show ephone offhook** command output. The **show voip rtp connections** command showed active call information in the system but it did not apply to ephone call legs. In Cisco Unified CME 8.8 and later versions, you can display information on active RTP calls, including the ephone tag number of the phone with an active call, the channel of the ephone-dn, and the caller and called party's numbers for the connection for both local and remote endpoints, using the **show ephone rtp connections** command. The output from this command provides an overview of all the connections in the system, narrowing the criteria for debugging pulse code modulation and Cisco Unified CME packets without a sniffer.

Note When an ephone to non-ephone call is made, information on the non-ephone does not appear in a **show ephone rtp connections** command output. To display the non-ephone call information, use the **show voip rtp connections** command.

The following sample output shows all the connected ephones in the Cisco Unified CME system. The sample output shows five active ephone connections with one of the phones having the **dspfarm-assist** keyword configured to transcode the code on the local leg to the indicated codec. The output also shows four ephone-to-ephone calls, represented in the CallID columns of both the RTP connection source and RTP connection destination by zero values.

Normally, a phone can have only one active connection but in the presence of a whisper intercom call, a phone can have two. In the sample output, ephone-40 has two active calls: it is receiving both a normal call and a whisper intercom call. The whisper intercom call is being sent by ephone-6, which has an invalid LocalIP of 0.0.0.0. The invalid LocalIP indicates that it does not receive RTP audio because it only has a one-way voice connection to the whisper intercom call recipient.

```
Router# show ephone rtp connections
Ephone RTP active connections :
Ephone    Line  DN Chan  SrcCallID  DstCallID            Codec (xcoded?)
    SrcNum DstNum  LocalIP                RemoteIP
ephone-5     1   5    1         15         14              G729 (Y)
    1005  1102  [192.168.1.100]:23192  [192.168.1.1]:2000
ephone-6     2  35    1          0          0       G711Ulaw64k (N)
    1035  1036  [0.0.0.0]:0  [192.168.1.81]:21256
ephone-40    1 140    1          0          0       G711Ulaw64k (N)
    1140  1141  [192.168.1.81]:21244  [192.168.1.70]:20664
ephone-40    2  36    1          0          0       G711Ulaw64k (N)
    1035  1036  [192.168.1.81]:21256  [192.168.1.1]:2000
ephone-41    1 141    1          0          0       G711Ulaw64k (N)
    1140  1141  [192.168.1.70]:20664  [192.168.1.81]:21244

Found 5 active ephone RTP connections
```

Ephone-Type Configuration

In Cisco Unified CME 4.3 and later versions, you can dynamically add a new phone type to your configuration without upgrading your Cisco IOS software. New phone models that do not introduce new features can easily be added to your configuration without requiring a software upgrade.

The ephone-type configuration template is a set of commands that describe the features supported by a type of phone, such as the particular phone type's device ID, number of buttons, and security support. Other phone-related settings under telephony-service, ephone-template, and ephone configuration mode can override the features set within the ephone-type template. For example, an ephone-type template can specify that a particular phone type supports security and another configuration setting can disable this feature. However, if an ephone-type template specifies that this phone does not support security, the other configuration cannot enable support for the security feature.

Cisco Unified CME uses the ephone-type template to generate XML files to provision the phone. System-defined phone types continue to be supported without using the ephone-type configuration. Cisco Unified CME checks the ephone-type against the system-defined phone types. If there is conflict with the phone type or the device ID, the configuration is rejected.

For configuration information, see the "SCCP: Configuring Ephone-Type Templates" section on page 230.

Support for 7926G Wireless SCCP IP Phone

Cisco Unified CME 8.6 adds support for the Cisco Unified 7926G Wireless SCCP IP phone. The 7926G wireless phone is phone similar to the 7925 wireless phone with a 2D barcode and EA15 module attached. The 7926G wireless phone is capable of scanning functionality. For more details on phone features and functionality, see *Cisco Unified IP Phone 7900 Series User Guide.*

Cisco Unified CME 8.6 supports the scanning function on the 7926G SCCP wireless phone using the ephone built-in device type. Table 21 shows supported values for the ephone-type for 7926G wireless phone.

Table 21 Supported Values for Ephone-Type Command

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified Wireless IP Phone 7926G	577	7926	6	2

To support service provisioning, an XML file is constructed externally and applied to the ephone-template of the phone. To allow the phone to read the external XML file, you are required to create-cnf and download the XML file to the ephone. For more information on configuring PhoneServices XML file, see the "SCCP: Configuring Phone Services XML File for Cisco Unified Wireless Phone 7926G" section on page 294.

The following is an example of the <phoneServices> XML file:

```
<phoneServices  useHTTPS="true">
<provisioning>0</provisioning>
<phoneService  type="1" category="0">
<name>Missed Calls</name>
<url>Application:Cisco/MissedCalls</url>
<vendor></vendor>
<version></version>
</phoneService>
<phoneService  type="0" category="1">
<displayName>Store Ops</displayName>
<name>Store Ops</name>
<url>http://1.4.206.105/Midlets/StoreOps.jad?StoreNumber=1777</url>
<http://1.4.206.105/Midlets/StoreOps.jad?StoreNumber=1777%3c/url%3e>
<http://1.4.206.105/Midlets/StoreOps.jad?StoreNumber=1777%3c/url%3e>
<vendor>CiscoSystems</vendor>
<version>0.0.82</version>
</phoneService>
</phoneServices>
```

KEM Support for Cisco Unified 8961, 9951, and 9971 SIP IP Phones

Cisco Unified IP Key Expansion Modules (KEMs) are supported on Cisco Unified 8961, 9951, and 9971 SIP IP phones in Cisco Unified CME 9.1.

You attach KEMs to supported phones to increase line key and feature key appearances, speed dials, or programmable buttons on your phones.

Table 22 lists the number of keys supported on Cisco Unified 8961, 9951, and 9971 SIP IP phones without KEMs.

Table 22 Number of Configurable Keys on Supported Cisco Unified SIP IP Phones Without KEMs

Number of Keys	8961	9951	9971
Fixed feature keys	5	5	6
Line keys	5	5	6
Programmable soft keys	5	5	6

With KEMs, the programmable buttons can be configured as phone line buttons, speed-dial buttons, or phone feature buttons.

Table 23 compares the number of feature keys that can be configured on supported Cisco Unified SIP IP phones with and without KEMs.

Table 23 Number of Configurable Feature Keys

Feature	Without KEMs	With KEMs
Busy-Lamp-Field speed dial	1 to 11	1 to 113
Directory number	1 to 12	1 to 114
Speed dial	1 to 11	1 to 113

Table 24 lists the maximum number of KEMs supported on Cisco Unified 8961, 9951, and 9971 SIP IP phones.

Table 24 Maximum Number of Supported KEMs and Additional Lines or Buttons

Cisco Unified SIP IP Phones	Maximum Number of Supported KEMs	Maximum Number of Additional Lines or Buttons
8961	1	36
9951	2	72
9971	3	108

Key Mapping

The mapping of configured keys on a phone depends on the number of KEMs attached to the phone.

If only one KEM is attached to a phone and the number of keys configured is 114, only 36 keys on the KEM are mapped to the configured keys on the phone. The rest of the keys are not visible on the phone or the KEM.

Call Control

All call control features are supported by KEMs on Cisco Unified 8961 SIP IP phones. Any feature that can be configured on the phone keys can also be configured on the KEM.

Because the Transfer, Hold, and Conference keys are built-in keys on Cisco Unified 9951 and 9971 SIP IP Phones, these features cannot be mapped to the keys on the KEMs.

XML Updates

- There is no separate firmware for KEMs, instead they are built in as part of the phones.
- The number of XML entries in the configuration file increases with the number of keys configured.
- The device type for KEMs is CKEM and the maximum number of supported keys is 36.

Restrictions

- KEMs are not supported for Cisco Unified SCCP IP phones and Cisco Unified SIP IP phones other than the Cisco Unified 8961, 9951, and 9971 SIP IP phones.
- Features configured on keys are disabled when supported Cisco Unified SIP IP phones are in Cisco Unified SIP SRST.
- All Cisco Unified 8961, 9951, and 9971 SIP IP phone restrictions and limitations apply to KEMs.
- All Cisco Unified CME and Cisco Unified SIP SRST feature restrictions and limitations apply to KEMs.

For more information on how the **blf-speed-dial**, **number**, and **speed-dial** commands, in voice register pool configuration mode, have been modified, see *Cisco Unified Communications Manager Express Command Reference*.

For information on installing KEMs on Cisco Unified IP Phone, see the "Installing a Key Expansion Module on the Cisco Unified IP Phone" section of *Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 7.1 (3) (SIP)*.

Fast-Track Configuration Approach for Cisco Unified SIP IP Phones

In Cisco Unified CME Release 10, the Fast-Track Configuration feature provides a new configuration utility using which you can input the phone characteristics of a new SIP phone model. This utility allows you to configure the existing SIP line features to the new SIP phone models. In the fast-track configuration, an option is provided to input an existing SIP phone as a reference phone. This feature is supported only on new SIP phone models that do not need any changes in the software protocols and the Cisco Unified CME application.

Note To deploy Cisco Unified SIP IP phones on Cisco Unified CME using the fast-track configuration approach, you require Cisco IOS Release 15.3(3)M or a later release.

Forward Compatibility

When a new SIP phone model is configured using the fast-track configuration approach. and the Cisco Unified CME is upgraded to a later version that supports the new SIP phone model, the fast-track configuration pertaining to that SIP phone model is removed automatically. If the Cisco Unified CME is downgraded to a version that does not have the built-in support, the fast-track configuration should be applied again.

To support Fast-Track Configuration feature, the **voice register pool-type** command has been introduced in the global configuration mode. The properties of the new SIP phone can be configured under the voice register pool-type submode. In addition to the explicit configuration of the phone's properties, the reference-pooltype option can be used to inherit the properties of an existing SIP phone.

Restrictions

- The fast-track configuration does not allow you to use the following phone models as reference phone:
  - ATA—Cisco ATA-186 and Cisco ATA-188
  - 7905—Cisco Unified IP Phone 7905 and Cisco Unified IP Phone 7905G

   - 7912—Cisco Unified IP Phone 7912 and Cisco Unified IP Phone 7912G
   - 7940—Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7940G
   - 7960—Cisco Unified IP Phone 7960 and Cisco Unified IP Phone 7960G
   - P100—PingTel Xpressa 100
   - P600—Polycom SoundPoint IP 600
- Existing Cisco Unified SIP IP phones are not allowed to be configured as new Cisco Unified SIP IP phones using the fast-track configuration approach.
- The reference-pooltype functionality is allowed only on existing SIP phone models. New SIP phone models configured using the fast-track configuration approach cannot be used as a reference phone.
- The fast-track configuration approach supports only the XML format and not support the text format for phone configuration.
- the fast-track approach does not support the new SIP phone models that have a new call flow, new message flow, or a new configuration file format that are not supported by the Cisco Unified CME.

For configuration information, see the "SIP: Provisioning Using the Fast-Track Configuration Approach" section on page 306.

For configuration examples, see the "Example: Fast-Track Configuration Approach" section on page 322.

How to Configure Phones for a PBX System

This section contains the following tasks:

- SCCP: Creating Directory Numbers, page 227 (required)
- SCCP: Configuring Ephone-Type Templates, page 230 (optional)
- SCCP: Assigning Directory Numbers to Phones, page 233 (required)
- SIP: Creating Directory Numbers, page 237 (required)
- SIP: Assigning Directory Numbers to Phones, page 240 (required)
- SIP: Configuring Dial Plans, page 243 (optional)
- SIP: Verifying Dial Plan Configuration, page 247 (optional)
- SIP: Enabling KPML, page 248 (optional)
- SIP: Selecting Session-Transport Protocol for a Phone, page 250 (optional)
- SIP: Disabling SIP Proxy Registration for a Directory Number, page 252 (required)
- Modifying the Global Codec, page 254
- Configuring Codecs of Individual Phones for Calls Between Local Phones, page 256 (required)

SCCP: Creating Directory Numbers

To create a directory number in Cisco Unified CME for a SCCP phone, intercom line, voice port, or a message-waiting indicator (MWI), perform the following steps for each directory number to be created. Each ephone-dn becomes a virtual line, or extension, on which call connections can be made. Each ephone-dn configuration automatically creates one or more virtual dial peers and virtual voice ports to make those call connections.

Note To create and assign directory numbers to be included in an overlay set, see the "SCCP: Configuring Overlaid Ephone-dns" section on page 909.

Prerequisites

- Maximum number of directory numbers must be changed from the default of 0 by using the **max-dn** command.
- Octo-line directory numbers are supported in Cisco Unified CME 4.3 and later versions.

Restrictions

- The Cisco Unified IP Phone 7931G is a SCCP keyset phone and, when configured for a key system, does not support the dual-line option for a directory number. To configure a Cisco Unified IP Phone 7931G, see the "How to Configure Phones for a Key System" section on page 258.
- Octo-line directory numbers are not supported by the Cisco Unified IP Phone 7902, 7920, or 7931, or by analog phones connected to the Cisco VG224 or Cisco ATA.
- Octo-line directory numbers are not supported in button overlay sets.
- Octo-line directory numbers do not support the **trunk** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag* [**dual-line** | **octo-line**]
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. **huntstop** [**channel** *number*]
6. **name** *name*
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	**ephone-dn** *dn-tag* [**dual-line** | **octo-line**] **Example:** `Router(config)# ephone-dn 7 octo-line`	Enters ephone-dn configuration mode to create a directory number for a SCCP phone. • **dual-line**—(Optional) Enables two calls per directory number. Supports features such as call waiting, call transfer, and conferencing with a single ephone-dn. • **octo-line**—(Optional) Enables eight calls per directory number. Supported in Cisco Unified CME 4.3 and later versions. • To change the line mode of a directory number, for example from dual-line to octo-line or the reverse, you must first delete the ephone-dn and then recreate it.
Step 4	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 2001`	Configures an extension number for this directory number. • Configuring a secondary number supports features such as call waiting, call transfer, and conferencing with a single ephone-dn.
Step 5	**huntstop** [**channel** *number*] **Example:** `Router(config-ephone-dn)# huntstop channel 4`	(Optional) Enables Channel Huntstop, which keeps a call from hunting to the next channel of a directory number if the first channel is busy or does not answer. • **channel** *number*—Number of channels available to accept incoming calls. Remaining channels are reserved for outgoing calls and features such as call transfer, call waiting, and conferencing. Range: 1 to 8. Default: 8. • *number* argument is supported for octo-line directory numbers only.
Step 6	**name** *name* **Example:** `Router(config-ephone-dn)# name Smith, John`	(Optional) Associates a name with this directory number. • Name is used for caller-ID displays and in the local directory listings. • Must follow the name order that is specified with the **directory** command.
Step 7	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

Examples

Nonshared Octo-Line Directory Number

In the following example, ephone-dn 7 is assigned to phone 10 and not shared by any other phone. There are two active calls on ephone-dn 7. Because the **busy-trigger-per-button** command is set to 2, a third incoming call to extension 2001 is either rejected with a busy tone or forwarded to another destination if Call Forward Busy is configured. The phone user can still make an outgoing call or transfer or conference a call on ephone-dn 7 because the **max-calls-per-button** command is set to 3, which allows a total of three calls on ephone-dn 7.

```
ephone-dn  7  octo-line
 number 2001
 name Smith, John
 huntstop channel 4
!
!
ephone  10
 max-calls-per-button 3
 busy-trigger-per-button 2
 mac-address 00E1.CB13.0395
 type 7960
 button  1:7
```

Shared Octo-Line Directory Number

In the following example, ephone-dn 7 is shared between phone 10 and phone 11. There are two active calls on ephone-dn 7. A third incoming call to ephone-dn 7 rings only phone 11 because its **busy-trigger-per-button** command is set to 3. Phone 10 allows a total of three calls, but it rejects the third incoming call because its **busy-trigger-per-button** command is set to 2. A fourth incoming call to ephone-dn 7 on ephone 11 is either rejected with a busy tone or forwarded to another destination if Call Forward Busy is configured. The phone user can still make an outgoing call or transfer or conference a call on ephone-dn 7 on phone 11 because the **max-calls-per-button** command is set to 4, which allows a total of four calls on ephone-dn 7 on phone 11.

```
ephone-dn  7  octo-line
 number 2001
 name Smith, John
 huntstop channel 4
!
!
ephone  10
 max-calls-per-button 3
 busy-trigger-per-button 2
 mac-address 00E1.CB13.0395
 type 7960
 button  1:7
!
!
!
ephone  11
 max-calls-per-button 4
 busy-trigger-per-button 3
 mac-address 0016.9DEF.1A70
 type 7960
 button  1:7
```

What to Do Next

After creating directory numbers, you can assign one or more directory numbers to a Cisco Unified IP Phone. See the "SCCP: Assigning Directory Numbers to Phones" section on page 233.

SCCP: Configuring Ephone-Type Templates

To add an IP phone type by defining an ephone-type template, perform the following steps.

Prerequisites

Cisco Unified CME 4.3 or a later version.

Restrictions

Ephone-type templates are not supported for system-defined phone types. For a list of system-defined phone types, see the **type** command in *Cisco Unified CME Command Reference.*

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-type** *phone-type* [**addon**]
4. **device-id** *number*
5. **device-name** *name*
6. **device-type** *phone-type*
7. **num-buttons** *number*
8. **max-presentation** *number*
9. **addon**
10. **security**
11. **phoneload**
12. **utf8**
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	**ephone-type** *phone-type* [**addon**] **Example:** `Router(config)# ephone-type E61`	Enters ephone-type configuration mode to create an ephone-type template. • *phone-type*—Unique label that identifies the type of IP phone for which the phone-type template is being defined. • **addon**—(Optional) Phone type is an add-on module, such as the Cisco Unified IP Phone 7915 Expansion Module.
Step 4	**device-id** *number* **Example:** `Router(config-ephone-type)# device-id 376`	Specifies the device ID for the phone type. • This device ID must match the predefined device ID for the specific phone model. • If this command is set to the default value of 0, the ephone-type is invalid. • See Table 25 for a list of supported device IDs.
Step 5	**device-name** *name* **Example:** `Router(config-ephone-type)# device-name E61 Mobile Phone`	Assigns a name to the phone type. • See Table 25 for a list of supported device types.
Step 6	**device-type** *phone-type* **Example:** `Router(config-ephone-type)# device-type E61`	Specifies the device type for the phone.
Step 7	**num-buttons** *number* **Example:** `Router(config-ephone-type)# num-buttons 1`	Number of line buttons supported by the phone type. • *number*—Range: 1 to 100. Default: 0. • See Table 25 for the number of buttons supported by each phone type.
Step 8	**max-presentation** *number* **Example:** `Router(config-ephone-type)# max-presentation 1`	Number of call presentation lines supported by the phone type. • *number*—Range: 1 to 100. Default: 0. • See Table 25 for the number of presentation lines supported by each phone type.

	Command or Action	Purpose
Step 9	**addon** **Example:** `Router(config-ephone-type)# addon`	(Optional) Specifies that this phone type supports an add-on module, such as the Cisco Unified IP Phone 7915 Expansion Module.
Step 10	**security** **Example:** `Router(config-ephone-type)# security`	(Optional) Specifies that this phone type supports security features. • This command is enabled by default.
Step 11	**phoneload** **Example:** `Router(config-ephone-type)# phoneload`	(Optional) Specifies that this phone type requires that the **load** command be configured. • This command is enabled by default.
Step 12	**utf8** **Example:** `Router(config-ephone-type)# utf8`	(Optional) Specifies that this phone type supports UTF8. • This command is enabled by default.
Step 13	**end** **Example:** `Router(config-ephone-type)# end`	Exits to privileged EXEC mode.

Ephone-Type Parameters for Supported Phone Types

Table 25 lists the required device ID, device type, and the maximum number of buttons and call presentation lines that are supported for each phone type that can be added with ephone-type templates.

Table 25 Supported Values for Ephone-Type Commands

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 6901	547	6901	1	1
Cisco Unified IP Phone 6911	548	6911	10	1
Cisco Unified IP Phone 6945	564	6945	4	2
Cisco Unified IP Phone 7915 Expansion Module with 12 buttons	227	7915	12	0 (default)
Cisco Unified IP Phone 7915 Expansion Module with 24 buttons	228	7915	24	0
Cisco Unified IP Phone 7916 Expansion Module with 12 buttons	229	7916	12	0
Cisco Unified IP Phone 7916 Expansion Module with 24 buttons	230	7916	24	0
Cisco Unified Wireless IP Phone 7925	484	7925	6	4
Cisco Unified IP Conference Station 7937G	431	7937	1	6
Cisco Unified IP Phone 8941	586	8941	4	3

Table 25 Supported Values for Ephone-Type Commands (continued)

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 8945	585	8945	4	3
Cisco Unified IP Phone 8941 with Fast-Track configuration support	586	8941	4	3
Cisco Unified IP Phone 8945 with Fast-Track configuration support	586	8945	4	3
Nokia E61	376	E61	1	1

Examples

The following example shows the Nokia E61 added with an ephone-type template, which is then assigned to ephone 2:

```
ephone-type  E61
 device-id 376
 device-name E61 Mobile Phone
 num-buttons 1
 max-presentation 1
 no utf8
 no phoneload
!
ephone  2
 mac-address 001C.821C.ED23
 type E61
 button  1:2
```

SCCP: Assigning Directory Numbers to Phones

This task sets up the initial ephone-dn-to-ephone relationships: how and which extensions appear on each phone. To create and modify phone-specific parameters for individual SCCP phones, perform the following steps for each SCCP phone to be connected in Cisco Unified CME. While using the GUI to administer ephone-dns on CME, ensure ephone-dns value is lower than the max-dns value.

Note To create and assign directory numbers to be included in an overlay set, see the “SCCP: Configuring Overlaid Ephone-dns” section on page 909.

Prerequisites

- To configure a phone line for Watch (w) mode by using the **button** command, Cisco Unified CME 4.1 or a later version.
- To configure a phone line for Monitor (m) mode by using the **button** command, Cisco CME 3.0 or a later version.
- To assign a user-defined phone type in Cisco Unified CME 4.3 or a later version, you must first create an ephone-type template. See the “SCCP: Configuring Ephone-Type Templates” section on page 230.

Restrictions

- For Watch mode. If the watched directory number is associated with several phones, then the watched phone is the one on which the watched directory number is on button 1 or the one on which the watched directory number is on the button that is configured by using the **auto-line** command, with auto-line having priority. For configuration information, see the "Configuring Automatic Line Selection" section on page 661.
- Octo-line directory numbers are not supported by the Cisco Unified IP Phone 7902, 7920, or 7931, or by analog phones connected to the Cisco VG224 or Cisco ATA.
- Octo-line directory numbers are not supported in button overlay sets.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone-tag*
4. **mac-address** [*mac-address*]
5. **type** *phone-type* [**addon 1** *module-type* [**2** *module-type*]]
6. **button** *button-number*{*separator*}*dn-tag* [**,***dn-tag*...] [*button-number*{**x**}*overlay-button-number*] [*button-number*...]
7. **max-calls-per-button** *number*
8. **busy-trigger-per-button** *number*
9. **keypad-normalize**
10. **nte-end-digit-delay** [*milliseconds*]
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	`ephone` *`phone-tag`* **Example:** `Router(config)# ephone 6`	Enters ephone configuration mode. • *phone-tag*—Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type **?** to display range.

	Command or Action	Purpose
Step 4	**mac-address** [*mac-address*] **Example:** `Router(config-ephone)# mac-address 2946.3f2.311`	Specifies the MAC address of the IP phone that is being configured. • *mac-address*—(Optional) For Cisco Unified CME 3.0 and later versions, it is not required to register phones before configuring the phone because Cisco Unified CME can detect MAC addresses and automatically populate phone configurations with the MAC addresses and phone types for individual phones. Not supported for voice-mail ports.
Step 5	**type** *phone-type* [**addon 1** *module-type* [**2** *module-type*]] **Example:** `Router(config-ephone)# type 7960 addon 1 7914`	Specifies the type of phone. • Cisco Unified CME 4.0 and later versions—The only types to which you can apply an add-on module are **7960**, **7961**, **7961GE**, and **7970**. • Cisco CME 3.4 and earlier versions—The only type to which you can apply an add-on module is **7960**.
Step 6	**button** *button-number*{*separator*}*dn-tag* [,*dn-tag*...] [*button-number*{**x**}*overlay-button-number*] [*button-number*...] **Example:** `Router(config-ephone)# button 1:10 2:11 3b12 4o13,14,15`	Associates a button number and line characteristics with an extension (ephone-dn). Maximum number of buttons is determined by phone type. **Note** The Cisco Unified IP Phone 7910 has only one line button but can be given two ephone-dn tags.
Step 7	**max-calls-per-button** *number* **Example:** `Router(config-ephone)# max-calls-per-button 3`	(Optional) Sets the maximum number of calls, incoming and outgoing, allowed on an octo-line directory number on this phone. • *number*—Range: 1 to 8. Default: 8. • This command is supported in Cisco Unified CME 4.3 and later versions. • This command must be set to a value that is more than or equal to the value set with the **busy-trigger-per-button** command. • This command can also be configured in ephone-template configuration mode and applied to one or more phones. The ephone configuration has priority over the ephone-template configuration.

	Command or Action	Purpose
Step 8	**busy-trigger-per-button** *number* **Example:** `Router(config-ephone)# busy-trigger-per-button 2`	(Optional) Sets the maximum number of calls allowed on this phone's octo-line directory numbers before triggering Call Forward Busy or a busy tone. • *number*—Range: 1 to 8. Default: 0 (disabled). • This command is supported in Cisco Unified CME 4.3 and later versions. • After the number of existing calls, incoming and outgoing, on an octo-line directory number exceeds the number of calls set with this command, the next incoming call to the directory number is forwarded to the Call Forward Busy destination if configured, or the call is rejected with a busy tone. • This command must be set to a value that is less than or equal to the value set with the **max-calls-per-button** command. • This command can also be configured in ephone-template configuration mode and applied to one or more phones. The ephone configuration has priority over the ephone-template configuration.
Step 9	**keypad-normalize** **Example:** `Router(config-ephone)# keypad-normalize`	(Optional) Imposes a 200-millisecond delay before each keypad message from an IP phone. • When used with the **nte-end-digit-delay** command, this command ensures that the delay configured for a dtmf-end event is always honored.
Step 10	**nte-end-digit-delay** [*milliseconds*] **Example:** `Router(config-ephone)# nte-end-digit-delay 150`	(Optional) Specifies the amount of time that each digit in the RTP NTE end event in an RFC 2833 packet is delayed before being sent. • This command is supported in Cisco Unified CME 4.3 and later versions. • *milliseconds*—length of delay. Range: 10 to 200. Default: 200. • To enable the delay, you must also configure the **dtmf-interworking rtp-nte** command in voice-service or dial-peer configuration mode. For information, see the "Enabling DTMF Integration Using RFC 2833" section on page 560. • This command can also be configured in ephone-template configuration mode. The value set in ephone configuration mode has priority over the value set in ephone-template mode.
Step 11	**end** **Example:** `Router(config-ephone)# end`	Returns to privileged EXEC mode.

Examples

The following example assigns extension 2225 in the Accounting Department to button 1 on ephone 2:

```
ephone-dn 25
 number 2225
 name Accounting

ephone 2
 mac-address 00E1.CB13.0395
 type 7960
 button 1:25
```

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- After configuring phones in Cisco Unified CME to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361.

SIP: Creating Directory Numbers

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

Prerequisites

- Cisco CME 3.4 or a later version.
- SIP shared-line directory numbers are supported in Cisco Unified CME 7.1 and later versions.
- **registrar server** command must be configured. For configuration information, see the "Enabling Calls in Your VoIP Network" section on page 90.
- In Cisco Unified CME 7.1 and later versions, the maximum number of directory numbers must be changed from the default of 0 by using the **max-dn** (voice register global) command. For configuration information, see the "SIP: Setting Up Cisco Unified CME" section on page 159.

Restrictions

- Maximum number of directory numbers supported by a router is version and platform dependent.
- Call Forward All, Presence, and message-waiting indication (MWI) features in Cisco Unified CME 4.1 and later versions require that SIP phones be configured with a directory number using the **dn** keyword with the **number** command; direct line numbers are not supported.
- SIP endpoints are not supported on H.323 trunks. SIP endpoints are supported on SIP trunks only.
- The Media Flow-around feature configured with the **media flow-around** command is not supported by Cisco Unified CME with SIP phones.
- SIP shared-line directory numbers are not supported by the Cisco Unified IP Phone 7902, 7920, 7931, 7940, or 7960, or by analog phones connected to the Cisco VG224 or Cisco ATA.
- SIP shared-line directory numbers cannot be members of hunt groups.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number** *number*
5. **shared-line** [**max-calls** *number-of-calls*]
6. **huntstop channel** *number-of-channels*
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 dn** *dn-tag* **Example:** `Router(config)# 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).
Step 4	**number** *number* **Example:** `Router(config-register-dn)# number 7001`	Defines a valid number for a directory number.
Step 5	**shared-line** [**max-calls** *number-of-calls*] **Example:** `Router(config-register-dn)# shared-line max-calls 6`	(Optional) Creates a shared-line directory number. • **max-calls** *number-of-calls*—(Optional) Maximum number of calls, both incoming and outgoing. Range: 2 to 16. Default: 2. • Must be set to a value that is more than or equal to the value set with the **busy-trigger-per-button** command. • This command is supported in Cisco Unified CME 7.1 and later versions.

	Command or Action	Purpose
Step 6	**huntstop channel** *number-of-channels* **Example:** `Router(config-register-dn)# huntstop channel 3`	(Optional) Enables Channel Huntstop, which keeps a call from hunting to the next channel of a directory number if the first channel is busy or does not answer. • *number-of-channels*—Number of channels available to accept incoming calls on the directory number. Remaining channels are reserved for outgoing calls and features, such as Call Transfer, Call Waiting, and Conferencing. Range: 1 to 50. Default: 0 (disabled). • This command is supported in Cisco Unified CME 7.1 and later versions.
Step 7	**end** **Example:** `Router(config-register-dn)# end`	Exits to privileged EXEC mode.

Examples

The following example shows directory number 24 configured as a shared line and assigned to phone 124 and phone 125:

```
voice register dn  24
 number 8124
 shared-line max-calls 6
!
voice register pool  124
 id mac 0017.E033.0284
 type 7965
 number 1 dn 24
!
voice register pool  125
 id mac 00E1.CB13.0395
 type 7965
 number 1 dn 24
```

SIP: Assigning Directory Numbers to Phones

This task sets up which extensions appear on each phone. To create and modify phone-specific parameters for individual SIP phones, perform the following steps for each SIP phone to be connected in Cisco Unified CME.

Note If your Cisco Unified CME system supports SCCP and 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. **id**{**network** *address* **mask** *mask* | **ip** *address* **mask** *mask* | **mac** *address*}
5. **type** *phone-type*
6. **number** *tag* **dn** *dn-tag*
7. **busy-trigger-per-button** *number-of-calls*
8. **username** *username* **password** *password*
9. **dtmf-relay** [**cisco-rtp**] [**rtp-nte**] [**sip-notify**]
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	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone.
Step 4	**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.

	Command or Action	Purpose
Step 5	**type** *phone-type* **Example:** `Router(config-register-pool)# type 7960-7940`	Defines a phone type for the SIP phone being configured.
Step 6	**number** *tag* **dn** *dn-tag* **Example:** `Router(config-register-pool)# number 1 dn 17`	Associates a directory number with the SIP phone being configured. • **dn** *dn-tag*—Identifies the directory number for this SIP phone as defined by the **voice register dn** command.
Step 7	**busy-trigger-per-button** *number-of-calls* **Example:** `Router(config-register-pool)# busy-trigger-per-button 2`	(Optional) Sets the maximum number of calls allowed on any of this phone's directory numbers before triggering Call Forward Busy or a busy tone. • *number-of-calls*—Maximum number of calls allowed before Cisco Unified CME forwards the next incoming call to the Call Forward Busy destination, if configured, or rejects the call with a busy tone. Range: 1 to 50. • This command is supported in Cisco Unified CME 7.1 and later versions.
Step 8	**username** *username* **password** *password* **Example:** `Router(config-register-pool)# username smith password 123zyx`	(Optional) Required only if authentication is enabled with the **authenticate** command. Creates an authentication credential. **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: Admin.
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 the SIP phone to relay DTMF tones. **Note** SIP phones natively support in-band DTMF relay as specified in RFC 2833.
Step 10	**end** **Example:** `Router(config-register-pool)# end`	Returns to privileged EXEC mode.

Examples

SIP Nonshared Line

In the following example, voice register dn 23 is assigned to phone 123. The fourth incoming call to extension 8123 is not presented to the phone because the **huntstop channel** command is set to 3. Because the **busy-trigger-per-button** command is set to 2 on phone 123 and Call Forward Busy is configured, the third incoming call to extension 8123 is forwarded to extension 8200.

```
voice register dn  23
 number 8123
 call-forward b2bua busy 8200
 huntstop channel 3
!
voice register pool  123
 busy-trigger-per-button 2
 id mac 0009.A3D4.1234
 type 7965
 number 1 dn 23
```

SIP Shared Line

In the following example, voice register dn 24 is shared by phones 124 and 125. The first two incoming calls to extension 8124 ring both phones. A third incoming call rings only phone 125 because its **busy-trigger-per-button** command is set to 3. The fourth incoming call to extension 8124 triggers Call Forward Busy because the busy trigger limit on all phones is exceeded.

```
voice register dn  24
 number 8124
 call-forward b2bua busy 8200
 shared-line max-calls 6
 huntstop channel 6
!
voice register pool  124
 busy-trigger-per-button 2
 id mac 0017.E033.0284
 type 7965
 number 1 dn 24
!
voice register pool  125
 busy-trigger-per-button 3
 id mac 00E1.CB13.0395
 type 7965
 number 1 dn 24
```

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- If you want to select the session-transport protocol for a SIP phone, see the "SIP: Selecting Session-Transport Protocol for a Phone" section on page 250.
- If you are finished configuring phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SIP: Generating Configuration Profiles for SIP Phones" section on page 363.

SIP: Configuring Dial Plans

Dial plans enable SIP phones to recognize digit strings dialed by users. After the phone recognizes a dial pattern, it automatically sends a SIP INVITE message to the Cisco Unified CME to initiate the call and does not require the user to press the Dial key or wait for the interdigit timeout. To define a dial plan for a SIP phone, perform the following steps.

Prerequisites

- Cisco Unified CME 4.1 or a later version.
- **mode cme** command must be enabled in Cisco Unified CME.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dialplan** *dialplan-tag*
4. **type** *phone-type*
5. **pattern** *tag string* [**button** *button-number*] [**timeout** *seconds*] [**user** {**ip** | **phone**}]
   or
   filename *filename*
6. **exit**
7. **voice register pool** *pool-tag*
8. **dialplan** *dialplan-tag*
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 dialplan` *dialplan-tag* **Example:** `Router(config)# voice register dialplan 1`	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.

	Command or Action	Purpose
Step 4	**type** *phone-type* **Example:** `Router(config-register-dialplan)# type 7905-7912`	Defines a phone type for the SIP dial plan. • **7905-7912**—Cisco Unified IP Phone 7905, 7905G, 7912, or 7912G. • **7940-7960-others**—Cisco Unified IP Phone 7911, 7940, 7940G, 7941, 7941GE, 7960, 7960G, 7961, 7961GE, 7970, or 7971. • The phone type specified with this command must match the type of phone for which the dial plan is used. If this phone type does not match the type assigned to the phone with the **type** command in voice register pool mode, the dial-plan configuration file is not generated. • You must enter this command before using the **pattern** or **filename** command in the next step.
Step 5	**pattern** *tag string* [**button** *button-number*] [**timeout** *seconds*] [**user** {**ip** | **phone**}] or **filename** *filename* **Example:** `Router(config-register-dialplan)# pattern 1 52...` or `Router(config-register-dialplan)# filename dialsip`	Defines a dial pattern for a SIP dial plan. • *tag*—Number that identifies the dial pattern. Range: 1 to 24. • *string*—Dial pattern, such as the area code, prefix, and first one or two digits of the telephone number, plus wildcard characters or dots (.) for the remainder of the dialed digits. • **button** *button-number*—(Optional) Button to which the dial pattern applies. • **timeout** *seconds*—(Optional) Time, in seconds, that the system waits before dialing the number entered by the user. Range: 0 to 30. To have the number dialed immediately, specify 0. If you do not use this parameter, the phone's default interdigit timeout value is used (10 seconds). • **user**—(Optional) Tag that automatically gets added to the dialed number. Do not use this keyword if Cisco Unified CME is the only SIP call agent. • **ip**—Uses the IP address of the user. • **phone**—Uses the phone number of the user. • Repeat this command for each pattern that you want to include in this dial plan. or Specifies a custom XML file that contains the dial patterns to use for the SIP dial plan. • You must load the custom XML file must into flash and the filename cannot include the .xml extension. • The **filename** command is not supported for the Cisco Unified IP Phone 7905 or 7912.

	Command or Action	Purpose
Step 6	**exit** **Example:** `Router(config-register-dialplan)# exit`	Exits dialplan configuration mode.
Step 7	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 4`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is version and platform-dependent; type **?** to display range. You can modify the upper limit for this argument by using the the **max-pool** command.
Step 8	**dialplan** *dialplan-tag* **Example:** `Router(config-register-pool)# dialplan 1`	Assigns a dial plan to a SIP phone. • *dialplan-tag*—Number that identifies the dial plan to use for this SIP phone. This is the number that was used with the **voice register dialplan** command in Step 3. Range: 1 to 24.
Step 9	**end** **Example:** `Router(config-register-global)# end`	Exits to privileged EXEC mode.

Examples

The following example shows the configuration for dial plan 1, which is assigned to SIP phone 1:

```
voice register dialplan  1
 type 7940-7960-others
 pattern 1 2... timeout 10 user ip
 pattern 2 1234 user ip button 4
 pattern 3 65...
 pattern 4 1...!
!
voice register pool  1
 id mac 0016.9DEF.1A70
 type 7961GE
 number 1 dn 1
 number 2 dn 2
 dialplan 1
 dtmf-relay rtp-nte
 codec g711ulaw
```

Troubleshooting Tips

If you create a dial plan by downloading a custom XML dial pattern file to flash and using the **filename** command, and the XML file contains an error, the dial plan might not work properly on a phone. We recommend creating a dial pattern file using the **pattern** command.

To remove a dial plan that was created using a custom XML file with the **filename** command, you must remove the dial plan from the phone, create a new configuration profile, and then use the **reset** command to reboot the phone. You can use the **restart** command after removing a dial plan from a phone only if the dial plan was created using the **pattern** command.

To use KPML if a matching dial plan is not found, when both a dial plan and KPML are enabled on a phone, you must configure a dial pattern with a single wildcard character (.) as the last pattern in the dial plan. For example:

```
voice register dialplan 10
 type 7940-7960-others
 pattern 1 66...
 pattern 2 91.......
 pattern 3 .
```

What to Do Next

If you are done modifying parameters for SIP phones, you must generate a new configuration profile and restart the phones. See the "Generating Configuration Files for Phones" section on page 359.

SIP: Verifying Dial Plan Configuration

Step 1 **show voice register dialplan** *tag*

This command displays the configuration information for a specific SIP dial plan.

```
Router# show voice register dialplan 1

Dialplan Tag 1
Config:
  Type is 7940-7960-others
  Pattern 1 is 2..., timeout is 10, user option is ip, button is default
  Pattern 2 is 1234, timeout is 0, user option is ip, button is 4
  Pattern 3 is 65..., timeout is 0, user option is phone, button is default
  Pattern 4 is 1..., timeout is 0, user option is phone, button is default
```

Step 2 **show voice register pool** *tag*

This command displays the dial plan assigned to a specific SIP phone.

```
Router# show voice register pool 29

Pool Tag 29
Config:
  Mac address is 0012.7F54.EDC6
  Number list 1 : DN 29
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  keep-conference is enabled
  dialplan tag is 1
  kpml signal is enabled
  service-control mechanism is not supported
.
.
.
```

Step 3 **show voice register template** *tag*

This command displays the dial plan assigned to a specific template.

```
Router# show voice register template 3

Temp Tag 3
Config:
  Attended Transfer is disabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
  DnD control is enabled
  Anonymous call block is disabled
  Voicemail is 62000, timeout 15
  Dialplan Tag is 1
  Transport type is tcp
```

SIP: Enabling KPML

To enable KPML digit collection on a SIP phone, perform the following steps.

Prerequisites

Cisco Unified CME 4.1 or a later version.

Restrictions

- This feature is supported only on Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.
- A dial plan assigned to a phone has priority over KPML.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **digit collect kpml**
5. **end**
6. **show voice register dial-peers**

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 4`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is version and platform-dependent; type **?** to display range. You can modify the upper limit for this argument by using the **max-pool** command.
Step 4	**`digit collect kpml`** **Example:** `Router(config-register-pool)# digit collect kpml`	Enables KPML digit collection for the SIP phone. **Note** This command is enabled by default for supported phones in Cisco Unified CME.

	Command or Action	Purpose
Step 5	**end** **Example:** `Router(config-register-pool)# end`	Exits to privileged EXEC mode.
Step 6	**show voice register dial-peers** **Example:** `Router# show voice register dial-peers`	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified CME SIP register, including the defined digit collection method.

What to Do Next

If you are done modifying parameters for SIP phones, you must generate a new configuration profile and restart the phones. See the "Generating Configuration Files for Phones" section on page 359.

SIP: Selecting Session-Transport Protocol for a Phone

To change the session-transport protocol for a SIP phone from the default of UDP to TCP, perform the following steps.

Prerequisites

- Cisco Unified CME 4.1 or a later version.
- Directory number must be assigned to SIP phone to which configuration is to be applied. For configuration information, see the "SIP: Assigning Directory Numbers to Phones" section on page 240.

Restrictions

- TCP is not supported as a session-transport protocol for the Cisco Unified IP Phone 7905, 7912, 7940, or 7960. If TCP is assigned to an unsupported phone, calls to that phone will not complete successfully. However, the phone can originate calls using UDP, although TCP has been assigned.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **session-transport** {**tcp** | **udp**}
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 3`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone in Cisco Unified CME.

	Command or Action	Purpose
Step 4	**session-transport** {**tcp** | **udp**} **Example:** `Router(config-register-pool)# session-transport tcp`	(Optional) Specifies the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME. • This command can also be configured in voice register template configuration mode and applied to one or more phones. The voice register pool configuration has priority over the voice register template configuration.
Step 5	**end** **Example:** `Router(config-register-pool)# end`	Exits voice register pool configuration mode and enters privileged EXEC mode.

What to Do Next

- If you want to disable SIP Proxy registration for an individual directory number, see the "SIP: Disabling SIP Proxy Registration for a Directory Number" section on page 252.
- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- If you are finished configuring phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SIP: Generating Configuration Profiles for SIP Phones" section on page 363

SIP: Disabling SIP Proxy Registration for a Directory Number

To prevent a particular directory number from registering with an external SIP proxy server, perform the following steps.

Prerequisites

- Cisco Unified CME 3.4 or a later version.
- Bulk registration is configured at system level. For configuration information, see the "Configuring Bulk Registration" section on page 142.

Restrictions

Phone numbers that are registered under a voice register dn must belong to a SIP phone that is registered in Cisco Unified CME.

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.
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.

Step 5	**no-reg** **Example:** `Router(config-register-dn)# no-reg`	Prevents directory number being configured from registering with an external proxy server.
Step 6	**end** **Example:** `Router(config-register-dn)# end`	Exits voice register dn configuration mode and enters privileged EXEC mode.

What to Do Next

- If you want to configure the G.722-64K codec for all calls through your Cisco Unified CME system, see the "Modifying the Global Codec" section on page 254.
- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- If you want to configure individual phones to support some codec other than the system-level codec or some codec other than the phone's native codec, see the "Codecs for Cisco Unified CME Phones" section on page 210.
- If you are finished configuring phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SIP: Generating Configuration Profiles for SIP Phones" section on page 363

Modifying the Global Codec

To change the global codec from the default (G.711ulaw) to G.722-64K for all calls through Cisco Unified CME, perform the following steps.

Prerequisites

Cisco Unified CME 4.3 or later versions.

Restrictions

If G.722-64K codec is configured globally and a phone does not support the codec, the fallback codec is G.711ulaw.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **codec** {**g711-ulaw** | **g722-64k**}
5. **service phone g722CodecSupport** {**0** | **1** | **2**}
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	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony service configuration mode to set parameters for SCCP and SIP phones in Cisco Unified CME.
Step 4	**codec** {**g711-ulaw** | **g722-64k**} **Example:** `Router(config-telephony)# codec g722-64k`	Specifies the preferred codec for phones in Cisco Unified CME. • Required only if you want to modify codec from the default (G.711ulaw) to G.722-64K.

	Command or Action	Purpose
Step 5	**service phone g722CodecSupport** {**0** | **1** | **2**} **Example:** `Router(config)# service phone g722CodecSupport 2`	Causes all phones to advertise the G.722-64K codec to Cisco Unified CME. • Required only if you configured the **codec g722-64k** command in telephony-service configuration mode. • **g722CodecSupport**—Default: 0, phone default set by manufacturer and equal to enabled or disabled. • Cisco phone firmware 8.2.1 or a later version is required to support the G.722-64K codec on G.722-capable SCCP phones. • Cisco phone firmware 8.3.1 or a later version is required to support the G.722-64K codec on G.722-capable SIP phones. • For SCCP only: This command can also be configured in ephone- template configuration mode and applied to one or more SCCP phones.
Step 6	**end** **Example:** `Router(config-telephony)# end`	Exits the telephony service configuration mode and enters privileged EXEC mode.

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- If you want to configure individual phones to support some codec other than the system-level codec or some codec other than the phone's native codec, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- If you are finished configuring SCCP phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361.

Configuring Codecs of Individual Phones for Calls Between Local Phones

To designate a codec for individual phones to ensure connectivity between a variety of phones connected to the same Cisco Unified CME router, perform the following steps for each SCCP or SIP phone.

Note If codec values for the dial peers of an internal connection do not match, the call fails. For calls to external phones, that is, phones that are not in the same Cisco Unified CME, such as VoIP calls, the codec is negotiated based on the protocol that is used for the call, such as H.323. Cisco Unified CME plays no part in the negotiation.

Prerequisites

- For SIP phones in Cisco Unified CME: Cisco Unified CME 3.4 or a later version.
- For G.722-64K and iLBC codecs: Cisco Unified CME 4.3 or a later version.
- To support G.722-64K on an individual phone: Cisco phone firmware 8.2.1 or a later version for SCCP phones and 8.3.1 or a later version for SIP phones. For information about upgrading Cisco phone firmware, see the "Installing and Upgrading Cisco Unified CME Software" section on page 61.
- To support iLBC on an individual phone: Cisco phone firmware 8.3.1 or a later version for SCCP and SIP phones. For information about upgrading Cisco phone firmware, see the "Installing and Upgrading Cisco Unified CME Software" section on page 61.
- Cisco Unified IP phone to which the codec is to be applied must be already configured. For configuration information for SIP phones, see the "SIP: Assigning Directory Numbers to Phones" section on page 240. For configuration information for SCCP phones, see the "SCCP: Assigning Directory Numbers to Phones" section on page 233.

Restrictions

- Not all phones support all codecs. To verify whether your phone supports a particular codec, see your phone documentation.
- For SIP and SCCP phones in Cisco Unified CME: Modify the configuration for either SIP or SCCP phones to ensure that the codec for all phones match. Do not modify the configuration for both SIP and SCCP phones.
- 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 second port of the Cisco ATA device, it will be disconnected gracefully. If you want to use both Cisco ATA ports simultaneously, then configure G.711 in Cisco Unified CME.
- If G.722-64K or iLBC codecs are configured in ephone configuration mode and the phone does not support the codec, the fallback is the global codec or G.711 ulaw if the global codec is not supported. To configure a global codec, see the "Modifying the Global Codec" section on page 254.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *ephone-tag*
   or
   voice register pool *pool-tag*
4. **codec** *codec-type*
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 **`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 1`	Enters ephone configuration mode to set phone-specific parameters for a SCCP phone in Cisco Unified CME. or Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone in Cisco Unified CME.
Step 4	**`codec`** *`codec-type`* **Example:** `Router(config-ephone)# codec g729r8` or `Router(config-register-pool)# codec g711alaw`	Specifies the codec for the dial peer for the IP phone being configured. • *codec-type*—Type **?** for a list of codecs. • This command overrides any previously configured codec selection set with the **voice-class codec** command. • This command overrides any previously configured codec selection set with the **codec** command in telephony-service configuration mode. • SCCP only—This command can also be configured in ephone-template configuration mode and applied to one or more phones.
Step 5	**`end`** **Example:** `Router(config-ephone)# end` or `Router(config-register-pool)# end`	Exits the configuration mode and enters privileged EXEC mode.

What to Do Next

- If you want to select the session-transport protocol for a SIP phone, see the "SIP: Selecting Session-Transport Protocol for a Phone" section on page 250.
- If you are finished configuring SIP phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SIP: Generating Configuration Profiles for SIP Phones" section on page 363.
- If you are finished configuring SCCP phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361.

How to Configure Phones for a Key System

This section contains the following tasks:

- SCCP: Creating Directory Numbers for a Simple Key System, page 258 (required)
- SCCP: Configuring Trunk Lines for a Key System, page 261 (required)
- SCCP: Configuring Individual IP Phones for Key System, page 270 (required)

SCCP: Creating Directory Numbers for a Simple Key System

To create a set of directory numbers with the same number to be associated with multiple line buttons on an IP phone and provide support for call waiting and call transfer on a key system phone, perform the following steps.

Restrictions

- Do not configure directory numbers for a key system for dual-line mode because this does not conform to the key system one-call-per-line button usage model for which the phone is designed.
- Provisioning support for the Cisco Unified IP Phone 7931 is available only in Cisco Unified CME 4.0(2) and later versions.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag*
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. **preference** *preference-order*
6. **no huntstop**
   or
   huntstop
7. **mwi-type** {**visual** | **audio** | **both**}
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	**ephone-dn** *dn-tag* **Example:** `Router(config)# ephone-dn 11`	Enters ephone-dn configuration mode to create a directory number.
Step 4	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 101`	Configures a valid phone or extension number for this directory number.
Step 5	**preference** *preference-order* *Example:* `Router(config-ephone-dn)# preference 1`	Sets dial-peer preference order for a directory number associated with a Cisco Unified IP phone. • Default: 0. • Increments the preference order for all subsequent instances within a set of ephone dns with the same number to be associated with a key system phone. That is, the first instance of the directory number is preference **0** by default and you must specify **1** for the second instance of the same number, **2** for the next, and so on. This allows you to create multiple buttons with the same number on an IP phone. • Required to support call waiting and call transfer on a key system phone.

	Command or Action	Purpose
Step 6	**no huntstop** or **huntstop** **Example:** `Router(config-ephone-dn)# no huntstop` or `Router(config-ephone-dn)# huntstop`	Explicitly enables call hunting behavior for a directory number. • Configure **no huntstop** for all instances, except the final instance, within a set of ephone dns with the same number to be associated with a key system phone. • Required to allow call hunting across multiple line buttons with the same number on an IP phone. or Disables call hunting behavior for a directory number. • Configure the **huntstop** command for the final instance within a set of ephone dns with the same number to be associated with a key system phone. • Required to limit the call hunting to a set of multiple line buttons with the same number on an IP phone.
Step 7	**mwi-type** {**visual** | **audio** | **both**} **Example:** `Router(config-ephone-dn)# mwi-type audible`	Specifies the type of MWI notification to be received. • This command is supported only by Cisco Unified IP Phone 7931s and Cisco Unified IP Phone 7911s. • This command can also be configured in ephone-dn-template configuration mode. The value set in ephone-dn configuration mode has priority over the value set in ephone-dn-template mode.
Step 8	**end** **Example:** `Router(config-ephone-dn)# end`	Exits to privileged EXEC mode.

Examples

The following example shows the configuration for six instances of directory number 101, assigned to the first six buttons of an IP phone:

```
ephone-dn 10
 number 101
 no huntstop

ephone-dn 11
 number 101
 preference 1
 no huntstop

ephone-dn 12
 number 101
 preference 2
 no huntstop

ephone-dn 13
 number 101
 preference 3
 no huntstop
```

```
ephone-dn 14
 number 101
 preference 4
 no huntstop

ephone-dn 15
 number 101
 preference 5

ephone 1
 mac-address 0001.2345.6789
 type 7931
 button 1:10 2:11 3:12 4:13 5:14 6:15
```

SCCP: Configuring Trunk Lines for a Key System

To set up trunk lines for your key system, perform only one of the following procedures:

- To only enable direct status monitoring of the FXO port on the line button of the IP phone, see the "SCCP: Configuring a Simple Key System Phone Trunk Line Configuration" section on page 261
- To enable direct status monitoring and allow transferred PSTN FXO line calls to be automatically recalled if the transfer target does not answer, see the "SCCP: Configuring an Advanced Key System Phone Trunk Line Configuration" section on page 265.

SCCP: Configuring a Simple Key System Phone Trunk Line Configuration

Perform the steps in this section to:

- Create directory numbers corresponding to each FXO line that allows phones to have shared or private lines connected directly to the PSTN.
- Enable direct status monitoring of the FXO port on the line button of the IP phone. The line button indicator, either a lamp or an icon depending on the phone, shows the in-use status of the FXO port during the duration of the call.

Prerequisites

- FXO port for a private line automatic ringdown (PLAR) off-premises extension (OPX) connection must be configured; for example:

```
voice-port 1/0/0
 connection plar-opx 801 <<----Private number
```

- Dial peers for FXO port must be configured; for example:

```
dial-peer voice 111 pots
 destination-pattern 811 <<----Trunk-tag
 port 1/0/0
```

Restrictions

- Directory number with a trunk line cannot be configured for call forward, busy, or no answer.
- Numbers entered after a trunk line is seized will not be displayed. Only the trunk tag is displayed on IP phones.
- Numbers entered after trunk line is seized will not appear in call history or call detail records (CDRs) of a Cisco Unified CME router. Only the trunk tag is logged for calls made from trunk lines.
- FXO trunk lines do not support the CFwdALL, Transfer, Pickup, GPickUp, Park, CallBack, and NewCall soft keys.
- FXO trunk lines do not support conference initiator dropoff.
- FXO trunk lines do not support on-hook redial. The phone user must explicitly select the FXO trunk line before pressing the Redial button.
- FXO trunk lines do not support call transfer to IP phones. However, the call initiator can conference an FXO line with an IP phone by pressing the Hold button, which leaves the FXO trunk line and IP phone connected. The conference initiator is unable to participate in the conference, but can place calls on other lines.
- FXO trunk lines do not support bulk speed dial.
- FXO port monitoring has the following restrictions:
  - Not supported before Cisco Unified CME 4.0.
  - Supported only for analog FXO loop-start and ground-start ports and T1/E1 FXO CAS ports. FXS loop-start and ground-start ports and PRI/BRI PSTN trunks are not supported.
  - Not supported for analog ports on the Cisco VG224 or Cisco ATA 180 Series.
  - T1 CAS DS0 group must be configured per time slot (cannot bundle more than one time slot into a ds0-group).
- Transfer recall and transfer-to button optimization are supported on dual-line directory numbers only in Cisco Unified CME 4.0 and later versions.
- Transfer-to button optimization is not supported for call forwarding, call-park recall, call pickup on hold, or call pickup at alert.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag*
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. **trunk** *trunk-tag* [**timeout** *seconds*] **monitor-port** *port*
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 51`	Enters ephone-dn configuration mode to create a directory number. • Configure this command in the default single line mode, without the **dual-line** keyword, when configuring a simple key system trunk line.
Step 4	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 801`	Configures a valid phone or extension number for this directory number.
Step 5	**trunk** *trunk-tag* [**timeout** *seconds*] **monitor-port** *port* **Example:** `Router(config-ephone-dn)# trunk 811 monitor-port 1/0/0`	Associates a directory number with an FXO port. • The **monitor-port** keyword is not supported before Cisco Unified CME 4.0. • The **monitor-port** keyword is not supported on directory numbers for analog ports on the Cisco VG224 or Cisco ATA 180 Series.
Step 6	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

Examples

The following example shows the configuration for six instances of directory number 101, assigned to the first six buttons of an IP phone, plus four PSTN line appearances that are assigned to buttons 7 to 10:

```
ephone-dn 10
 number 101
 no huntstop

ephone-dn 11
 number 101
 preference 1
 no huntstop

ephone-dn 12
 number 101
 preference 2
 no huntstop
```

```
ephone-dn 13
 number 101
 preference 3
 no huntstop

ephone-dn 14
 number 101
 preference 4
 no huntstop

ephone-dn 15
 number 101
 preference 5

ephone-dn 51
 number 801
 trunk 811 monitor-port 1/0/0

ephone-dn 52
 number 802
 trunk 812 monitor-port 1/0/1

ephone-dn 53
 number 803
 trunk 813 monitor-port 1/0/2

ephone-dn 54
 number 804
 trunk 814 monitor-port 1/0/3

ephone 1
 mac-address 0001.2345.6789
 type 7931
 button 1:11 2:12 3:13 4:14 5:15 6:16 7:51 8:52 9:53 10:54

voice-port 1/0/0
 connection plar opx 801

voice-port 1/0/1
 connection plar opx 802

voice-port 1/0/2
 connection plar opx 803

voice-port 1/0/3
 connection plar opx 804

dial-peer voice 811 pots
 destination-pattern 811
 port 1/0/0

dial-peer voice 812 pots
 destination-pattern 812
 port 1/0/1

dial-peer voice 813 pots
 destination-pattern 813
 port 1/0/2

dial-peer voice 814 pots
 destination-pattern 814
 port 1/0/3
```

What to Do Next

You are ready to configure each individual phone and assign button numbers, line characteristics, and directory numbers to buttons on the phone. See the "SCCP: Configuring Individual IP Phones for Key System" section on page 270.

SCCP: Configuring an Advanced Key System Phone Trunk Line Configuration

Perform the steps in this section to:

- Create directory numbers corresponding to each FXO line that allows phones to have shared or private lines connected directly to the PSTN.
- Enable direct status monitoring of the FXO port on the line button of the IP phone. The line button indicator, either a lamp or an icon depending on the phone, shows the in-use status of the FXO port during the duration of the call.
- Allow transferred PSTN FXO line calls to be automatically recalled if the transfer target does not answer after the specified number of seconds. The call is withdrawn from the transfer-to phone and the call resumes ringing on the phone that initiated the transfer.

Prerequisites

- FXO port for a private line automatic ringdown (PLAR) off-premises extension (OPX) connection must be configured; for example:

```
voice-port 1/0/0
 connection plar-opx 801 <<----Private number
```

- Dial peers for FXO port must be configured; for example:

```
dial-peer voice 111 pots
 destination-pattern 811 <<----Trunk-tag
 port 1/0/0
```

Restrictions

- Ephone-dn with a trunk line cannot be configured for call forward, busy, or no answer.
- Numbers entered after a trunk line is seized will not be displayed. Only the trunk tag is displayed on IP phones.
- Numbers entered after a trunk line is seized will not appear in call history or call detail records (CDRs) of a Cisco Unified CME router. Only the trunk tag is logged for calls made from trunk lines.
- FXO trunk lines do not support the CFwdALL, Transfer, Pickup, GPickUp, Park, CallBack, and NewCall soft keys.
- FXO trunk lines do not support conference initiator dropoff.
- FXO trunk lines do not support on-hook redial. The phone user must explicitly select the FXO trunk line before pressing the Redial button.
- FXO trunk lines do not support call transfer to IP phones. However, the call initiator can conference an FXO line with an IP phone by pressing the Hold button, which leaves the FXO trunk line and IP phone connected. The conference initiator is unable to participate in the conference, but can place calls on other lines.
- FXO trunk lines do not support bulk speed dial.

- FXO port monitoring has the following restrictions:
  - Not supported before Cisco Unified CME 4.0.
  - Supported only for analog FXO loop-start and ground-start ports and T1/E1 FXO CAS ports. FXS loop-start and ground-start ports and PRI/BRI PSTN trunks are not supported.
  - Not supported for analog ports on the Cisco VG224 or Cisco ATA 180 Series.
  - T1 CAS DS0 group must be configured per time slot (cannot bundle more than one time slot into a ds0-group).
- Transfer recall and transfer-to button optimization is supported on dual-line directory numbers only in Cisco Unified CME 4.0 and later.
- Transfer-to button optimization is not supported for call forwarding, call-park recall, call pickup on hold, or call pickup at alert.
- Transfer recall is not supported for analog ports on the Cisco VG224 or Cisco ATA 180 Series.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag* **dual-line**
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. **trunk** *digit-string* [**timeout** *seconds*] [**transfer-timeout** *seconds*] [**monitor-port** *port*]
6. **huntstop** [**channel**]
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.

	Command or Action	Purpose
Step 3	**ephone-dn** *dn-tag* **dual-line** **Example:** `Router(config)# ephone-dn 51 dual-line`	Enters ephone-dn configuration mode for the purpose of creating and configuring a telephone or extension number. • **dual-line**—Required when configuring an advanced key system phone trunk line. Dual-line mode provides a second call channel for the directory number on which to place an outbound consultation call during the call transfer attempt. This also allows the phone to remain part of the call to monitor the progress of the transfer attempt and if the transfer is not answered, to pull the call back to the phone on the original PSTN line button.
Step 4	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 801`	Configures a valid telephone number or extension number for this directory number.
Step 5	**trunk** *digit-string* [**timeout** *seconds*] [**transfer-timeout** *seconds*] [**monitor-port** *port*] **Example:** `Router(config-ephone-dn)# trunk 811 transfer-timeout 30 monitor-port 1/0/0`	Associates this directory number with an FXO port. • **transfer-timeout** *seconds*—For dual-line ephone-dns only. Range: 5 to 60000. Default: Disabled. • The **monitor-port** keyword is not supported before Cisco Unified CME 4.0. • The **monitor-port** and **transfer-timeout** keywords are not supported on directory numbers for analog ports on the Cisco VG224 or Cisco ATA 180 Series.
Step 6	**huntstop** [**channel**] **Example:** `Router(config-ephone-dn)# huntstop channel`	Disables call hunting to the second channel of this directory number if the first channel is busy or does not answer. • **channel**—Required when configuring an advanced key system phone trunk line. Reserves the second channel created by configuring dual-line mode for the **ephone-dn** command so that an outbound consultation call can be placed during a call transfer attempt.
Step 7	**end** **Example:** `Router(config-ephone-dn)# end`	Exits to privileged EXEC mode.

Examples

The following example shows the configuration for six instances of directory number 101, assigned to the first six buttons of an IP phone, plus four PSTN line appearances that are assigned to buttons 7 to 10. These four PSTN line appearances are configured as dual lines to provide a second call channel on which to place an outbound consultation call during a call transfer attempt. This configuration allows the phone to remain part of the call to monitor the progress of the transfer attempt, and if the transfer is not answered, to pull the call back to the phone on the original PSTN line button.

```
ephone-dn 10
 number 101
 no huntstop

ephone-dn 11
 number 101
 preference 1
 no huntstop

ephone-dn 12
 number 101
 preference 2
 no huntstop

ephone-dn 13
 number 101
 preference 3
 no huntstop

ephone-dn 14
 number 101
 preference 4
 no huntstop

ephone-dn 15
 number 101
 preference 5

ephone-dn 51 dual-line
 number 801
 trunk 811 transfer-timeout 30 monitor-port 1/0/0
 huntstop channel

ephone-dn 52 dual-line
 number 802
 trunk 812 transfer-timeout 30 monitor-port 1/0/1
 huntstop channel

ephone-dn 53 dual-line
 number 803
 trunk 813 transfer-timeout 30 monitor-port 1/0/2
 huntstop channel

ephone-dn 54 dual-line
 number 804
 trunk 814 transfer-timeout 30 monitor-port 1/0/3
 huntstop channel

ephone 1
 mac-address 0001.2345.6789
 type 7931
 button 1:11 2:12 3:13 4:14 5:15 6:16 7:51 8:52 9:53 10:54

voice-port 1/0/0
```

```
 connection plar opx 801

voice-port 1/0/1
 connection plar opx 802

voice-port 1/0/2
 connection plar opx 803

voice-port 1/0/3
 connection plar opx 804

dial-peer voice 811 pots
 destination-pattern 811
 port 1/0/0

dial-peer voice 812 pots
 destination-pattern 812
 port 1/0/1

dial-peer voice 813 pots
 destination-pattern 813
 port 1/0/2

dial-peer voice 814 pots
 destination-pattern 814
 port 1/0/3
```

SCCP: Configuring Individual IP Phones for Key System

To assign button numbers, line characteristics, and directory numbers to buttons on an individual phone that will operate as a key system phone, perform the following steps.

Restrictions

- Provisioning for Cisco Unified IP Phone 7931G is available only in Cisco Unified CME 4.0(2) and later versions.
- Cisco Unified IP Phone 7931G can support only one call waiting overlaid per directory number.
- Cisco Unified IP Phone 7931G cannot support overlays that contain directory numbers configured for dual-line mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone-tag*
4. **mac-address** [*mac-address*]
5. **type** *phone-type*
6. **button** *button-number*{*separator*}*dn-tag* [**,***dn-tag*...] [*button-number*{**x**}*overlay-button-number*] [*button-number*...]
7. **mwi-line** *line-number*
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	**ephone** *phone-tag* **Example:** `Router(config)# ephone 1`	Enters ephone configuration mode.
Step 4	**mac-address** [*mac-address*] **Example:** `Router(config-ephone)# mac-address 0001.2345.6789`	Specifies the MAC address of the IP phone that is being configured.

	Command or Action	Purpose
Step 5	**type** *phone-type* **Example:** `Router(config-ephone)# type 7931`	Specifies the type of phone that is being configured.
Step 6	**button** *button-number*{*separator*}*dn-tag* [**,***dn-tag*...] [*button-number*{**x**}*overlay-button-number*] [*button-number*...] **Example:** `Router(config-ephone)# button 1:11 2:12 3:13 4:14 5:15 6:16 7:51 8:52 9:53 10:54`	Associates a button number and line characteristics with an ephone-dn. Maximum number of buttons is determined by phone type. **Tip** The line button layout for the Cisco Unified IP Phone 7931G is a bottom-up array. Button 1 is at the bottom right of the array and button 24 is at the top left of the array.
Step 7	**mwi-line** *line-number* **Example:** `Router(config-ephone)# mwi-line 3`	Selects a phone line to receive MWI treatment; when a message is waiting for the selected line, the message waiting indicator is activated. • *line-number*—Range: 1 to 34. Default: 1.
Step 8	**end** **Example:** `Router(config-ephone)# end`	Exits ephone configuration mode and enters privileged EXEC mode.

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- To select a fixed-button layout for a Cisco Unified IP Phone 7931G, see the "SCCP: Selecting Button Layout for a Cisco Unified IP Phone 7931G" section on page 1476.
- If you are finished configuring phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361.

How to Configure Cisco ATA, Analog Phone Support, Remote Phones, Cisco IP Communicator, and Secure IP Phone (IP-STE)

This section contains the following tasks:

Cisco ATA

- Configuring Cisco ATA Support, page 272 (required)
- Verifying Cisco ATA Support, page 274 (optional)
- Using Call Pickup and Group Call Pickup with Cisco ATA, page 276 (optional)
- Configuring Voice and T.38 Fax Relay on Cisco ATA-187, page 277 (optional)

Analog Phones

- SCCP: Enabling Auto-Configuration for Cisco VG202, VG204, and VG224, page 281
- SCCP: Configuring Phones on SCCP Controlled Analog (FXS) Ports, page 284 (required)
- SCCP: Verifying Analog Phone Support, page 287 (optional)

Remote phones

- SCCP: Enabling a Remote Phone, page 287 (required)
- SCCP: Verifying Remote Phones, page 289 (optional)

Cisco IP Communicator

- SCCP: Configuring Cisco IP Communicator Support, page 289 (required)
- SCCP: Verifying Cisco IP Communicator Support, page 290 (required)
- SCCP: Troubleshooting Cisco IP Communicator Support, page 291 (optional)

Secure IP Phones

- SCCP: Configuring Secure IP Phone (IP-STE), page 292

Cisco Unified Wireless Phone 7926G

- SCCP: Configuring Phone Services XML File for Cisco Unified Wireless Phone 7926G, page 294 (required)

Configuring Cisco ATA Support

To enable an analog phone that uses a Cisco ATA to register with Cisco Unified CME, perform the following steps.

Restrictions

For a Cisco ATA that is registered to a Cisco Unified CME system to participate in fax calls, it must have its ConnectMode parameter set to use the same RTP payload type as the Cisco voice gateway that is performing the fax pass-through. Cisco voice gateways use standard payload type 0/8, which is selected on Cisco ATAs by setting bit 2 of the ConnectMode parameter to 1. For more information, see the "Parameters and Defaults" chapter in *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP (version 3.0)*.

SUMMARY STEPS

1. Install Cisco ATA.
2. Configure Cisco ATA for SCCP.
3. Upgrade firmware.
4. Set network parameters on Cisco ATA.
5. Configure analog phones in Cisco Unified CME.

DETAILED STEPS

Step 1 Install the Cisco ATA.

See the "Installing the Cisco ATA" chapter in *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP (version 3.0)*.

Step 2 Configure the Cisco ATA.

See the "Configuring the Cisco ATA for SCCP" chapter in *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP (version 3.0).*

Step 3 Upgrade the firmware to the latest Cisco ATA image.

If you are using either the v2.14 or v2.14ms Cisco ATA 186 image based on the 2.14 020315a build for H.323/SIP or the 2.14 020415a build for MGCP or SCCP, you must upgrade to the latest version to install a security patch. This patch fixes a security hole in the Cisco ATA Web server that allows users to bypass the user interface password.

For information about upgrading firmware, see the "Installing and Upgrading Cisco Unified CME Software" section on page 61. Alternatively, you can use a manual method, as described in the "Upgrading the Cisco ATA Signaling Image" chapter of *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP (version 3.0)*.

Step 4 Set the following network parameters on the Cisco ATA:

- DHCP parameter to **1** (enabled).
- TFTP parameter to **1** (enabled).
- TFTPURL parameter to the IP address of the router running Cisco Unified CME.
- SID0 parameter to a period (**.**) or the MAC address of the Cisco ATA (to enable the first port).
- SID1 parameter to a period (**.**) or a modified version the Cisco ATA's MAC address, with the first two hexadecimal numbers removed and 01 appended to the end, if you want to use the second port. For example, if the MAC address of the Cisco ATA is 00012D01073D, set SID1 to 012D01073D01.
- Nprintf parameter to the IP address and port number of the host to which all Cisco ATA debug messages are sent. The port number is usually set to 9001.
- To prevent tampering and unauthorized access to the Cisco ATA 186, you can disable the web-based configuration. However, if you disable the web configuration page, you must use either a TFTP server or the voice configuration menu to configure the Cisco ATA 186.

Step 5 In Cisco Unified CME, configure analog phones that use a Cisco ATA in the same way as a Cisco Unified IP phone. In the **type** command, use the **ata** keyword. For information on how to provision phones, see the "SCCP: Creating Directory Numbers" section on page 227.

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- To select a fixed-button layout for a Cisco Unified IP Phone 7931G, see the "SCCP: Selecting Button Layout for a Cisco Unified IP Phone 7931G" section on page 1476.
- If you are finished configuring phones to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361 and the "SIP: Generating Configuration Profiles for SIP Phones" section on page 363.

Verifying Cisco ATA Support

Use the **show ephone ata** command to display SCCP phone configurations with the **type ata** command.

The following is sample output for a Cisco Unified CME configured for two analog phones using a Cisco ATA with MAC address 000F.F758.E70E:

```
ephone-30 Mac:000F.F758.E70E TCP socket:[2] activeLine:0 REGISTERED in SCCP ver 1 and
Server in ver 1
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:7
IP:1.4.188.72 15325 ATA Phone  keepalive 7 max_line 2 dual-line
button 1: dn 80 number 8080 CH1   IDLE         CH2   IDLE

ephone-31 Mac:0FF7.58E7.0E01 TCP socket:[3] activeLine:0 REGISTERED in SCCP ver 1 and
Server in ver 1
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:3
IP:1.4.188.72 15400 ATA Phone  keepalive 7 max_line 2 dual-line
button 1: dn 81 number 8081 CH1   IDLE         CH2   IDLE
```

Troubleshooting Cisco ATA Support

Use the **debug ephone detail** command to diagnose problems with analog phones that use Cisco ATAs.

The following is sample output for two analog phones using a Cisco ATA with MAC address 000F.F758.E70E. The sample shows the activities that take place when the phones register.

```
Router# debug ephone detail mac-address 000F.F758.E70E

*Apr  5 02:50:11.966: New Skinny socket accepted [1] (33 active)
*Apr  5 02:50:11.970: sin_family 2, sin_port 15325, in_addr 1.4.188.72
*Apr  5 02:50:11.970: skinny_add_socket 1 1.4.188.72 15325
21:21:49: %IPPHONE-6-REG_ALARM: Name=ATA000FF758E70E Load=ATA030203SCCP051201A.zup
Last=Initialized
*Apr  5 02:50:11.974:
Skinny StationAlarmMessage on socket [2] 1.4.188.72 ATA000FF758E70E
*Apr  5 02:50:11.974: severityInformational p1=0 [0x0] p2=0 [0x0]
*Apr  5 02:50:11.974: Name=ATA000FF758E70E Load=ATA030203SCCP051201A.zup Last=Initialized
*Apr  5 02:50:12.066: ephone-(30)[2] StationRegisterMessage (29/31/48) from 1.4.188.72
*Apr  5 02:50:12.066: ephone-(30)[2] Register StationIdentifier DeviceName ATA000FF758E70E
*Apr  5 02:50:12.070: ephone-(30)[2] StationIdentifier Instance 1    deviceType 12
*Apr  5 02:50:12.070: ephone-30[-1]:stationIpAddr 1.4.188.72
*Apr  5 02:50:12.070: ephone-30[-1]:maxStreams 0
*Apr  5 02:50:12.070: ephone-30[-1]:protocol Ver 0x1
*Apr  5 02:50:12.070: ephone-30[-1]:phone-size 5392 dn-size 632
*Apr  5 02:50:12.070: ephone-(30) Allow any Skinny Server IP address 1.4.188.65
*Apr  5 02:50:12.070: ephone-30[-1]:Found entry 29 for 000FF758E70E
```

```
*Apr  5 02:50:12.070: ephone-30[-1]:socket change -1 to 2
*Apr  5 02:50:12.070: ephone-30[-1]:FAILED: CLOSED old socket -1
*Apr  5 02:50:12.074: ephone-30[2]:phone ATA000FF758E70E re-associate OK on socket [2]
21:21:49: %IPPHONE-6-REGISTER: ephone-30:ATA000FF758E70E IP:1.4.188.72 Socket:2
DeviceType:Phone has registered.
*Apr  5 02:50:12.074: Phone 29 socket 2
*Apr  5 02:50:12.074: Phone 29 socket 2: Running Bravo ??
*Apr  5 02:50:12.074: Skinny Local IP address = 1.4.188.65 on port 2000

*Apr  5 02:50:12.074: Skinny Phone IP address = 1.4.188.72 15325
*Apr  5 02:50:12.074: ephone-30[2]:Signal protocol ver 8 to phone with ver 1
*Apr  5 02:50:12.074: ephone-30[2]:Date Format M/D/Y
*Apr  5 02:50:12.078: ephone-30[2]:RegisterAck sent to ephone 2: keepalive period 30 use
sccp-version 1
*Apr  5 02:50:12.078: ephone-30[2]:CapabilitiesReq sent
*Apr  5 02:50:12.090: ephone-30[2]:VersionReq received
*Apr  5 02:50:12.090: ephone-30[2]:Version String not needed for ATA device. Part of XML
file
*Apr  5 02:50:12.090: ephone-30[2]:Version Message sent
*Apr  5 02:50:12.094: ephone-30[2]:CapabilitiesRes received
*Apr  5 02:50:12.098: ephone-30[2]:Caps list 7
G711Ulaw64k  60 ms
G711Alaw64k  60 ms
G729  60 ms
G729AnnexA  60 ms
G729AnnexB  60 ms
G729AnnexAwAnnexB  60 ms
Unrecognized Media Type 257  60 ms

*Apr  5 02:50:12.098: ephone-30[2]:ButtonTemplateReqMessage
*Apr  5 02:50:12.098: ephone-30[2]:StationButtonTemplateReqMessage set max presentation
to 2
*Apr  5 02:50:12.098: ephone-30[2]:CheckAutoReg
*Apr  5 02:50:12.102: ephone-30[2]:AutoReg is disabled
*Apr  5 02:50:12.102: ephone-30[2][ATA000FF758E70E]:Setting 1 lines 4 speed-dials on phone
(max_line 2)
*Apr  5 02:50:12.102: ephone-30[2]:First Speed Dial Button location is 2 (0)
*Apr  5 02:50:12.102: ephone-30[2]:Configured 4 speed dial buttons
*Apr  5 02:50:12.102: ephone-30[2]:ButtonTemplate lines=1 speed=4 buttons=5 offset=0
*Apr  5 02:50:12.102: ephone-30[2]:Skinny IP port 16384 set for socket [2]
*Apr  5 02:50:12.126: ephone-30[2]:StationSoftKeyTemplateReqMessage
*Apr  5 02:50:12.126: ephone-30[2]:StationSoftKeyTemplateResMessage
*Apr  5 02:50:12.206: ephone-30[2]:StationSoftKeySetReqMessage
*Apr  5 02:50:12.206: ephone-30[2]:StationSoftKeySetResMessage
*Apr  5 02:50:12.307: ephone-30[2]:StationLineStatReqMessage from ephone line 1
*Apr  5 02:50:12.307: ephone-30[2]:StationLineStatReqMessage ephone line 1 DN 80 = 8080
desc = 8080 label =
*Apr  5 02:50:12.307: ephone-30[2][ATA000FF758E70E]:StationLineStatResMessage sent to
ephone (1 of 2)
*Apr  5 02:50:12.427: ephone-30[2]:StationSpeedDialStatReqMessage speed 9
*Apr  5 02:50:12.427: ephone-30[2]:No speed-dial set 9
*Apr  5 02:50:12.427: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:12.547: ephone-30[2]:StationSpeedDialStatReqMessage speed 8
*Apr  5 02:50:12.547: ephone-30[2]:No speed-dial set 8
*Apr  5 02:50:12.547: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:12.635: ephone-30[2]:StationSpeedDialStatReqMessage speed 7
*Apr  5 02:50:12.635: ephone-30[2]:No speed-dial set 7
*Apr  5 02:50:12.635: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:12.707: New Skinny socket accepted [1] (34 active)
*Apr  5 02:50:12.707: sin_family 2, sin_port 15400, in_addr 1.4.188.72
*Apr  5 02:50:12.711: skinny_add_socket 1 1.4.188.72 15400
*Apr  5 02:50:12.711: ephone-30[2]:StationSpeedDialStatReqMessage speed 6
*Apr  5 02:50:12.711: ephone-30[2]:No speed-dial set 6
*Apr  5 02:50:12.715: ephone-30[2]:StationSpeedDialStatMessage sent
```

```
21:21:50: %IPPHONE-6-REG_ALARM: Name=ATA0FF758E70E01 Load=ATA030203SCCP051201A.zup
Last=Initialized
*Apr  5 02:50:12.715:
Skinny StationAlarmMessage on socket [3] 1.4.188.72 ATA000FF758E70E
*Apr  5 02:50:12.715: severityInformational p1=0 [0x0] p2=0 [0x0]
*Apr  5 02:50:12.715: Name=ATA0FF758E70E01 Load=ATA030203SCCP051201A.zup Last=Initialized
*Apr  5 02:50:12.811: ephone-30[2]:StationSpeedDialStatReqMessage speed 5
*Apr  5 02:50:12.811: ephone-30[2]:No speed-dial set 5
*Apr  5 02:50:12.811: ephone-30[2]:StationSpeedDialStatMessage sent
21:21:50: %IPPHONE-6-REGISTER: ephone-31:ATA0FF758E70E01 IP:1.4.188.72 Socket:3
DeviceType:Phone has registered.
*Apr  5 02:50:12.908: ephone-30[2]:StationSpeedDialStatReqMessage speed 4
*Apr  5 02:50:12.908: ephone-30[2]:No speed-dial set 4
*Apr  5 02:50:12.908: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:13.008: ephone-30[2]:StationSpeedDialStatReqMessage speed 3
*Apr  5 02:50:13.008: ephone-30[2]:No speed-dial set 3
*Apr  5 02:50:13.008: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:13.108: ephone-30[2]:StationSpeedDialStatReqMessage speed 2
*Apr  5 02:50:13.108: ephone-30[2]:No speed-dial set 2
*Apr  5 02:50:13.108: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:13.208: ephone-30[2]:StationSpeedDialStatReqMessage speed 1
*Apr  5 02:50:13.208: ephone-30[2]:No speed-dial set 1
*Apr  5 02:50:13.208: ephone-30[2]:StationSpeedDialStatMessage sent
*Apr  5 02:50:14.626: New Skinny socket accepted [1] (33 active)
*Apr  5 02:50:14.626: sin_family 2, sin_port 15593, in_addr 1.4.188.72
*Apr  5 02:50:14.630: skinny_add_socket 1 1.4.188.72 15593
*Apr  5 02:50:15.628: New Skinny socket accepted [1] (34 active)
*Apr  5 02:50:15.628: sin_family 2, sin_port 15693, in_addr 1.4.188.72
*Apr  5 02:50:15.628: skinny_add_socket 1 1.4.188.72 15693
*Apr  5 02:50:21.538: ephone-30[2]:SkinnyCompleteRegistration
```

Using Call Pickup and Group Call Pickup with Cisco ATA

Most of the procedures for using Cisco ATAs with Cisco Unified CME are the same as those for using Cisco ATAs with Cisco Unified Communications Manager, as described in the "How to Use Pre-Call and Mid-Call Services" chapter of *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP (version 3.0)*. However, the call pickup and group call pickup procedures are different when using Cisco ATAs with Cisco Unified CME, as described below:

Call Pickup

When using Cisco ATAs with Cisco Unified CME:

- To pickup the last parked call, press ****3***.
- To pickup a call on a specific extension, press ****3** and enter the extension number.
- To pickup a call from a park slot, press ****3** and enter the park slot number.

Group Call Pickup

When using Cisco ATAs with Cisco Unified CME:

- To answer a phone within your call pickup group, press ****4***.
- To answer a phone outside of your call pickup group, press ****4** and the group ID number.

Note If there is only one pickup group, you do not need to enter the group ID after the ****4** to pickup a call.

Configuring Voice and T.38 Fax Relay on Cisco ATA-187

To configure voice and T.38 Fax Relay on Cisco ATA-187, perform the following steps.

Prerequisites

Cisco Unified CME 9.0 or a later version.

Restrictions

- H.323 trunk calls are not supported.
- Hardware conferencing with DSPFarm resource is not supported on Cisco ATA-187 in Cisco Unified CME 9.0. With the correct firmware (9.2(3) or a later version), local three-way conferencing is supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **authenticate realm** *string*
5. **exit**
6. **voice service** {**voip** | **voatm**}
7. **allow-connections** *from-type* **to** *to-type*
8. **fax protocol t38** [**ls_redundancy** *value* [**hs_redundancy** *value*]] [**fallback** {**cisco** | **none** | **pass-through** {**g711ulaw** | **g711alaw**}}]
9. **exit**
10. **voice register pool** *pool-tag*
11. **id mac** *address*
12. **type** *phone-type*
13. **ata-ivr-pwd** *password*
14. **session-transport** {**tcp** | **udp**}
15. **number** *tag* **dn** *dn-tag*
16. **username** *username* [**password** *password*]
17. **codec** *codec-type* [*bytes*]
18. **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.
Step 4	**authenticate realm** *string* **Example:** `Router(config-register-global)# authenticate realm xxxxx`	• **realm** *string*—Realm parameter for challenge and response as specified in RFC 2617 is authenticated.
Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 6	**voice service** {**voip** | **voatm**} **Example:** `Router(config)# voice service voip`	Enters voice-service configuration mode to specify a voice encapsulation type. • **voip**—Specifies Voice over IP (VoIP) parameters. • **voatm**—Specifies Voice over ATM (VoATM) parameters.
Step 7	**allow-connections** *from-type* **to** *to-type* **Example:** `Router(config-voi-serv)# allow-connections sip to sip`	Allows connections between specific types of endpoints in a VoIP network. • *from-type*—Originating endpoint type. The following choices are valid: – **sip**—Session Interface Protocol. • **to**—Indicates that the argument that follows is the connection target. • *to-type*—Terminating endpoint type. The following choices are valid: – **sip**—Session Interface Protocol.

	Command or Action	Purpose
Step 8	**fax protocol t38** [**ls_redundancy** *value* [**hs_redundancy** *value*]] [**fallback** {**cisco** | **none** | **pass-through** {**g711ulaw** | **g711alaw**}}] **Example:** `Router(config-voi-serv)# fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw`	Specifies the global default ITU-T T.38 standard fax protocol to be used for all VoIP dial peers. • **ls_redundancy** *value*— (Optional) (T.38 fax relay only) Specifies the number of redundant T.38 fax packets to be sent for the low-speed V.21-based T.30 fax machine protocol. Range varies by platform from 0 (no redundancy) to 5 or 7. Default is 0. • **hs_redundancy** *value*— (Optional) (T.38 fax relay only) Specifies the number of redundant T.38 fax packets to be sent for high-speed V.17, V.27, and V.29 T.4 or T.6 fax machine image data. Range varies by platform from 0 (no redundancy) to 2 or 3. Default is 0. • **fallback**—(Optional) A fallback mode is used to transfer a fax across a VoIP network if T.38 fax relay could not be successfully negotiated at the time of the fax transfer. • **pass-through**—(Optional) The fax stream uses one of the following high-bandwidth codecs: – **g711ulaw**—Uses the G.711 u-law codec. – **g711alaw**—Uses the G.711 a-law codec.
Step 9	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits voice-service configuration mode.
Step 10	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool  11`	Enters voice register pool configuration mode to set phone-specific parameters for a Cisco Unified SIP phone in Cisco Unified CME. • *pool-tag*—Unique number assigned to the pool. Range: 1 to 100.
Step 11	**id mac** *address* **Example:** `Router(config-register-pool)# id mac 93FE.12D8.2301`	Identifies a locally available Cisco Unified SIP IP phone. • **mac** *address*—Identifies the MAC address of a particular Cisco Unified SIP IP phone.
Step 12	**type** *phone-type* **Example:** `Router(config-register-pool)# type ATA-187`	Defines a phone type for the SIP phone being configured.
Step 13	**ata-ivr-pwd** *password* **Example:** `Router(config-register-pool)# ata-ivr-pwd 1234`	(Optional) Defines a password to access interactive voice response (IVR) and change the default phone settings on Cisco Analog Telephone Adaptors. • *password*—Four-digit or five-digit string to be used as password to access IVR. Password string must contain numbers 0 to 9.

	Command or Action	Purpose
Step 14	**session-transport** {**tcp** | **udp**} **Example:** `Router(config-register-pool)# session-transport tcp`	(Optional) Specifies the transport layer protocol that a Cisco Unified SIP IP phone uses to connect to Cisco Unified CME. • **tcp**—Transmission Control Protocol (TCP) is used. • **udp**—User Datagram Protocol (UDP) is used. This is the default.
Step 15	**number** *tag* **dn** *dn-tag* **Example:** `Router(config-register-pool)# number 1 dn 33`	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from the Cisco Unified SIP IP phone. • *tag*—Identifies the telephone number when there are multiple number commands. Range: 1 to 10. • **dn** *dn-tag*—Identifies the directory number tag for this phone number as defined by the voice register dn command. Range: 1 to 150.
Step 16	**username** *username* [**password** *password*] **Example:** `Router(config-register-pool)# username ata112 password cisco`	Assigns an authentication credential to a phone user so that the SIP phone can register in Cisco Unified CME. • *username*—Username of the local Cisco IP phone user. Default: Admin. • **password**—Enables password for the Cisco IP phone user. • *password*—Password string.
Step 17	**codec** *codec-type* [*bytes*] **Example:** `Router(config-register-pool)# codec g711ulaw`	Specifies the codec to be used when setting up a call for a SIP phone or group of SIP phones in Cisco Unified CME. • *codec-type*—Preferred codec; values are as follows: – **g711alaw**—G.711 A–law 64K bps. – **g711ulaw**—G.711 micro–law 64K bps. – **g722r64**—G.722-64K at 64K bps. – **g729r8**—G.729 8K bps (default). – **ilbc**—internet Low Bitrate Codec (iLBC) at 13,330 bps or 15,200 bps.
Step 18	**end** **Example:** `Router(config-register-pool)# end`	Exits to privileged EXEC mode.

SCCP: Enabling Auto-Configuration for Cisco VG202, VG204, and VG224

To use the Autoconfiguration feature for voice gateways, perform the following steps on the Cisco Unified CME router.

Prerequisites

- Cisco Unified CME 7.1 or a later version. The Cisco Unified CME router must be configured and running before you boot the analog voice gateway. See the "SCCP: Setting Up Cisco Unified CME" section on page 146.
- Default location of configuration files is system:/its/. To define an alternate location at which to save the gateway configuration files, see the "SCCP: Defining Per-Phone Configuration Files and Alternate Location" section on page 152.
- To automatically assign the next available directory number to the voice port as it registers to Cisco Unified CME, and create an ephone entry associated with each voice port, enable the **auto assign** command in Cisco Unified CME.

Restrictions

Supported only for the Cisco VG202, VG204, and VG224 voice gateways.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-gateway system** *tag*
4. **mac-address** *mac-address*
5. **type** {**vg202** | **vg204** | **vg224**}
6. **voice-port** *port-range*
7. **network-locale** *locale-code*
8. **create cnf-file**
9. **reset**
   or
   restart
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	**voice-gateway system** *tag* **Example:** `Router(config)# voice-gateway system 1`	Enters voice gateway configuration mode and creates a voice gateway configuration.
Step 4	**mac-address** *mac-address* **Example:** `Router(config-voice-gateway)# mac-address`	Defines the MAC address of the voice gateway to autoconfigure.
Step 5	**type** {**vg202** | **vg204** | **vg224**} **Example:** `Router(config-voice-gateway)# type vg224`	Defines the type of voice gateway to autoconfigure.
Step 6	**voice-port** *port-range* **Example:** `Router(config-voice-gateway)# voice-port 0-23`	Identifies the ports on the voice gateway that register to Cisco Unified CME.
Step 7	**network-locale** *locale-code* **Example:** `Router(config-voice-gateway)# network-locale FR`	Selects a geographically specific set of tones and cadences for the voice gateway's analog endpoints that register to Cisco Unified CME.
Step 8	**create cnf-files** **Example:** `Router(config-voice-gateway)# create cnf-files`	Generates the XML configuration files that are required for the voice gateway to autoconfigure its analog ports that register to Cisco Unified CME.

	Command or Action	Purpose
Step 9	**reset** or **restart** **Example:** `Router(config-voice-gateway)# reset` or `Router(config-voice-gateway)# restart`	(Optional) Performs a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME. or (Optional) Performs a fast restart of all analog phones associated with the voice gateway after simple changes to buttons, lines, or speed-dial numbers. • Use these commands to download new configuration files to the analog phones after making configuration changes to the phones in Cisco Unified CME.
Step 10	**end** **Example:** `Router(config-voice-gateway)# end`	Exits to privileged EXEC mode.

Examples

The following example shows the voice gateway configuration in Cisco Unified CME:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

What to Do Next

- Cisco VG202 or VG204 voice gateway—Enable the gateway for autoconfiguration. See the "Auto-Configuration on the Cisco VG202 and Cisco VG204 Voice Gateways" section in *Cisco VG202 and Cisco VG204 Voice Gateways Software Configuration Guide*.
- Cisco VG224 analog phone gateway—Enable SCCP and the STC application on the gateway. See the "Configuring FXS Ports for Basic Calls" chapter in *Supplementary Services Features for FXS Ports on Cisco IOS Voice Gateways Configuration Guide*.

SCCP: Configuring Phones on SCCP Controlled Analog (FXS) Ports

Configuring Cisco Unified CME to support calls and features on analog endpoints connected to SCCP controlled analog (FXS) ports is basically the same as configuring any SCCP phone in Cisco Unified CME. This section describes only the steps that have special meaning for phones connected to a Cisco VG224 Analog Phone Gateway.

Prerequisites

- For phones connected to analog FXS ports on the Cisco VG224 Analog Phone Gateway: Cisco CME 3.2.2 or a later version.
- For phones connected to analog FXS ports on the Cisco Integrated Services Routers (ISR) voice gateway: Cisco Unified CME 4.0 or a later version.
- Cisco ISR voice gateway or Cisco VG224 analog phone gateway is installed and configured for operation. For information, see the appropriate Cisco configuration documentation.
- Prior to Cisco IOS Release 12.4(11)T, set the **timeouts ringing** command to **infinity** for all SCCP-controlled analog ports. In Cisco IOS Release 12.4(11)T and later, the default for this command is infinity.
- SCCP is enabled on the Cisco IOS voice gateway. For configuration information, see the *Supplementary Services Features for FXS Ports on Cisco IOS Voice Gateways Configuration Guide*.

Restrictions

FXS ports on Cisco VG248 analog phone gateways are not supported by Cisco Unified CME.

SUMMARY STEPS

1. Set up ephone-dns for up to 24 analog endpoints on the Cisco IOS gateway.
2. Set the maximum number of ephones.
3. Assign ephone-dns to ephones.
4. Set up feature parameters as desired.
5. Set up feature restrictions as desired.

DETAILED STEPS

Step 1 Set up ephone-dns for up to 24 endpoints on the Cisco IOS gateway.

Use the **ephone-dn** command:

```
ephone-dn 1 dual-line
 number 1000
.
.
.
ephone-dn 24 dual-line
 number 1024
```

Step 2 Set the maximum number of ephones.

Use the **max ephones** command to set a number equal to or greater than the total number of endpoints that you intend to register on the Cisco Unified CME router, including both IP and analog endpoints. For example, if you have 6 IP phones and 12 analog phones, set the **max ephones** command to 18 or greater.

Step 3 Assign ephone-dns to ephones.

Use the **auto assign** command to enable the automatic assignment of an available ephone-dn to each phone as the phone contacts the Cisco Unified CME router to register.

Note The order of ephone-dn assignment is not guaranteed. For example, if you have analog endpoints on ports 2/0 through 2/23 on the Cisco IOS gateway, port 2/0 does not necessarily become ephone 1. Use one of the following commands to enable automatic ephone-dn assignment.

- **auto assign 1 to 24**—You do not need to use the **type** keyword if you have only analog endpoints to be assigned or if you want all endpoints to be automatically assigned.
- **auto assign 1 to 24 type anl**—Use the **type** keyword if you have other phone types in the system and you want only the analog endpoints to be assigned to ephone-dns automatically.

An alternative to using the **auto assign** command is to manually assign ephone-dns to ephones (analog phones on FXS ports). This method is more complicated, but you might need to use it if you want to assign a specific extension number (ephone-dn) to a particular ephone. The reason that manual assignment is more complicated is because a unique device ID is required for each registering ephone and analog phones do not have unique MAC addresses like IP phones do. To create unique device IDs for analog phones, the auto assign process uses a particular algorithm. When you make manual ephone assignments, you have to use the same algorithm for each phone that receives a manual assignment.

The algorithm uses the single 12-digit SCCP local interface MAC address on the Cisco IOS gateway as the base to create unique 12-digit device IDs for all the FXS ports on the Cisco IOS gateway. The rightmost 9 digits of the SCCP local interface MAC address are shifted left three places and are used as the leftmost 9 digits for all 24 individual device IDs. The remaining 3 digits are the hexadecimal translation of the binary representation of the port's slot number (3 digits), subunit number (2 digits), and port number (7 digits). The following example shows the use of the algorithm to create a unique device ID for one port:

a. The MAC address for the Cisco VG224 SCCP local interface is 000C.8638.5EA6.

b. The FXS port has a slot number of 2 (010), a subunit number of 0 (00), and a port number of 1 (0000001). The binary digits are strung together to become 0100 0000 0001, which is then translated to 401 in hexadecimal to create the final device ID for the port and ephone.

c. The resulting unique device ID for this port is C863.85EA.6401.

When manually setting up an ephone configuration for an analog port, assign it just one button because the port represents a single-line device. The **button** command can use the ":" (colon, for normal), "o" (overlay) and "c" (call-waiting overlay) modes.

Note Once you have assigned ephone-dns to all the ephones that you want to assign manually, you can use the **auto assign** command to automatically assign the remaining ports.

Step 4 Set up feature parameters as desired.

The following list includes commonly configured features. For information about supported features, see *Supplementary Services Features for FXS Ports on Cisco IOS Voice Gateways Configuration Guide*.

- Call transfer—To use call transfer from analog endpoints, the **transfer-system** command must be configured for the **full-blind** or **full-consult** keyword in telephony-service configuration mode on the Cisco Unified CME router. This is the recommended setting for Cisco CME 3.0 and later versions, but it is not the default.
- Call forwarding—Call forwarding destinations are specified for all, busy, and no-answer conditions for each ephone-dn using the **call-forward all**, **call-forward busy**, and **call-forward noan** commands in ephone-dn configuration mode.
- Call park—Call-park slots are created using the **park-slot** command in ephone-dn configuration mode. Phone users must be instructed how to transfer calls to the call-park slots and use directed pickup to retrieve the calls.
- Call pickup groups—Extensions are added to pickup groups using the **pickup-group** command in ephone-dn configuration mode. Phone users must be told which phones are in which groups.
- Caller ID—Caller names are defined using the **name** command in ephone-dn configuration mode. Caller numbers are defined using the **number** command in ephone-dn configuration mode.
- Speed dial—Numbers to be speed-dialed are stored with their associated speed-dial codes using the **speed-dial** command in ephone configuration mode.
- Speed dial to voice mail—The voice-mail number is defined using the **voicemail** command in telephony-service configuration mode.

Step 5 Set up feature restrictions as desired.

Features such as transfer, conference, park, pickup, group pickup (gpickup), and call forward all (cfwdall) can be restricted from individual ephones using the appropriate Cisco Unified CME softkey template command, even though analog phones do not have soft keys. Simply create a template that leaves out the soft key that represents the feature you want to restrict and apply the template to the ephone for which you want the feature restricted. For more information about soft-key template customization, see the "Customizing Soft Keys" section on page 1335.

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- To select a fixed-button layout for a Cisco Unified IP Phone 7931G, see the "SCCP: Selecting Button Layout for a Cisco Unified IP Phone 7931G" section on page 1476.
- After configuring phones in Cisco Unified CME to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361.

SCCP: Verifying Analog Phone Support

Use the following **show** commands to display information about analog endpoints.

- **show ephone anl**—Displays MAC address, registration status, ephone-dn, and speed-dial numbers for analog ephones.
- **show telephony-service ephone-dn**—Displays call forward, call waiting, pickup group, and more information about ephone-dns.
- **show running-config**—Displays running configuration nondefault values.

SCCP: Enabling a Remote Phone

To enable IP phones or instances of Cisco IP Communicator to connect to a Cisco Unified CME system over a WAN, perform the following steps.

Prerequisites

- The WAN link supporting remote teleworker phones should be configured with a Call Admission Control (CAC) or Resource Reservation Protocol (RSVP) solution to prevent the oversubscription of bandwidth, which can degrade the quality of all voice calls.
- If DSP farms will be used for transcoding, you must configure them separately. See the "Configuring Transcoding Resources" section on page 449.
- A SCCP phone to be enabled as a remote phone is configured in Cisco Unified CME. For configuration information, see the "SCCP: Creating Directory Numbers" section on page 227.

Restrictions

- Because Cisco Unified CME is not designed for centralized call processing, remote phones are supported only for fixed teleworker applications, such as working from a home office.
- Cisco Unified CME does not support CAC for remote SCCP phones, so voice quality can degrade if a WAN link is oversubscribed. High-bandwidth data applications used over a WAN can cause degradation of voice quality for remote IP phones.
- Cisco Unified CME does not support Emergency 911 (E911) calls from remote IP phones. Teleworkers using remote phones connected to Cisco Unified CME over a WAN should be advised not to use these phones for E911 emergency services because the local public safety answering point (PSAP) will not be able to obtain valid calling-party information from them.

  We recommend that you make all remote phone users aware of this issue. One way is to place a label on all remote teleworker phones that reminds users not to place 911 emergency calls on remote IP phones. Remote workers should place any emergency calls through locally configured hotel, office, or home phones (normal land-line phones) whenever possible. Inform remote workers that if they must use remote IP phones for emergency calls, they should be prepared to provide specific location information to the answering PSAP personnel, including street address, city, state, and country.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone-tag*
4. **mtp**
5. **codec** {**g711ulaw** | **g722r64** | **g729r8** [**dspfarm-assist**]}
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** *phone-tag* **Example:** `Router(config)# ephone 36`	Enters ephone configuration mode. • *phone-tag*—Unique sequence number that identifies this ephone during configuration tasks.
Step 4	**mtp** **Example:** `Router(config-ephone)# mtp`	Sends media packets to the Cisco Unified CME router.
Step 5	**codec** {**g711ulaw** | **g722r64** | **g729r8** [**dspfarm-assist**]} **Example:** `Router(config-ephone)# codec g729r8 dspfarm-assist`	(Optional) Selects a preferred codec for setting up calls. • Default: G.711 mu-law codec. • The **g722r64** keyword requires Cisco Unified CME 4.3 and later versions. • **dspfarm-assist**—Attempts to use DSP-farm resources for transcoding the segment between the phone and the Cisco Unified CME router if G.711 is negotiated for the call. **Note** The **dspfarm-assist** keyword is ignored if the SCCP endpoint type is ATA, VG224, or VG248.
Step 6	**end** **Example:** `Router(config-ephone)# end`	Returns to privileged EXEC mode.

What to Do Next

- If you have SIP and SCCP phones connected to the same Cisco Unified CME, see the "Configuring Codecs of Individual Phones for Calls Between Local Phones" section on page 256.
- To select a fixed-button layout for a Cisco Unified IP Phone 7931G, see the "SCCP: Selecting Button Layout for a Cisco Unified IP Phone 7931G" section on page 1476.
- After configuring phones in Cisco Unified CME to make basic calls, you are ready to generate configuration files for the phones to be connected. See the "SCCP: Generating Configuration Files for SCCP Phones" section on page 361.

SCCP: Verifying Remote Phones

Step 1 Use the **show running-config** command or the **show telephony-service ephone** command to verify parameter settings for remote ephones.

SCCP: Configuring Cisco IP Communicator Support

To enable support for Cisco IP Communicator, perform the following steps.

Prerequisites

- Cisco Unified CME 4.0 or a later version.
- IP address of the Cisco Unified CME TFTP server.
- PC for Cisco IP Communicator is installed. For hardware and platform requirements, see the appropriate *Cisco IP Communicator User Guide*.
- Audio devices, such as headsets and handsets for users, are installed. You can install audio devices any time, but the ideal time to do this is before you install and launch Cisco IP Communicator.
- Directory numbers and ephone configuration for Cisco IP Communicator are configured in Cisco Unified CME. For information, see the "How to Configure Phones for a PBX System" section on page 226.

SUMMARY STEPS

1. Download Cisco IP Communicator 2.0 or a later version software.
2. Install and launch Cisco IP Communicator.
3. Complete the configuration and registration tasks on the Cisco IP Communicator as required, including:
   a. Configure IP address of the Cisco Unified CME TFTP server.
   b. Disable the Optimize for low bandwidth parameter.
4. Wait for Cisco IP Communicator to register.
5. Test Cisco IP Communicator.

DETAILED STEPS

Step 1 Download Cisco IP Communicator 2.0 or a later version software from the software download site at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp.

Step 2 Install the software on your PC, then launch the Cisco IP Communicator application.

For information, see the "Installing and Launching Cisco IP Communicator" section in the appropriate *Cisco IP Communicator User Guide*.

Step 3 Complete the configuration and registration tasks on the Cisco IP Communicator as required, including the following:

a. Configure the IP address of the Cisco Unified CME TFTP server.

- Right-click on the Cisco IP Communicator interface, then choose **Preferences** > **Network** > **Use these TFTP servers**.
- Enter the IP address of the Cisco Unified CME TFTP server in the field.

b. Disable the Optimize for low bandwidth parameter to ensure that Cisco IP Communicator sends voice packets for all calls.

Note The following steps are required to enable Cisco IP Communicator to support the G.711 codec, which is the fallback codec for Cisco Unified CME. You can compensate for disabling the optimization parameter by using the **codec** command in ephone configuration mode to configure G.729 or another advanced codec as the preferred codec for Cisco IP Communicator. This helps to ensure that the codec for a VoIP (For example, SIP or H.323) dial-peer is supported by Cisco IP Communicator and can prevent audio problems caused by insufficient bandwidth.

- Right-click on the Cisco IP Communicator interface and choose **Preferences > Audio**.
- Uncheck the checkbox next to Optimize for low bandwidth.

Step 4 Wait for the Cisco IP Communicator application to connect and register to Cisco Unified CME.

Step 5 Test Cisco IP Communicator.

For more information, see the "SCCP: Verifying Cisco IP Communicator Support" section on page 290.

SCCP: Verifying Cisco IP Communicator Support

Step 1 Use the **show running-config** command to display ephone-dn and ephone information associated with this phone.

Step 2 After Cisco IP Communicator registers with Cisco Unified CME, it displays the phone extensions and soft keys in its configuration. Verify that these are correct.

Step 3 Make a local call from the phone and have someone call you. Verify that you have a two-way voice path.

SCCP: Troubleshooting Cisco IP Communicator Support

Step 1 Use the **debug ephone detail** command to diagnose problems with calls. For more information, see *Cisco Unified CME Command Reference*.

SCCP: Configuring Secure IP Phone (IP-STE)

To configure an IP-STE phone on Cisco Unified CME, perform the following steps.

Prerequisites

Cisco Unified CME 8.0 or a later version.

Restrictions

- Detection or conversion between Network Transmission Equipment (NTE) and Session Signaling Event (SSE) is not supported.
- Transcoding or trans-compress rate support for different Voice Band Data (VBD) and Modem Relay (MR) media type is not supported.
- IP-STE supports only single-line calls, dual-line and octo-line calls are not supported.
- Speed-dial can only be configured manually on the IP-STE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone-tag*
4. **mac-address** [*mac-address*]
5. **type ip-ste**
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** *phone-tag* **Example:** `Router(config)# ephone 6`	Enters ephone configuration mode. • *phone-tag*—Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type **?** to display range.

	Command or Action	Purpose
Step 4	**mac-address** [*mac-address*] **Example:** `Router(config-ephone)# mac-address 2946.3f2.311`	Specifies the MAC address of the IP phone that is being configured.
Step 5	**type ip-ste** **Example:** `Router(config-ephone)# type ip-ste`	Specifies the type of phone.
Step 6	**end** **Example:** `Router(config-ephone)# end`	Returns to privileged EXEC mode.

SCCP: Configuring Phone Services XML File for Cisco Unified Wireless Phone 7926G

To configure the phone services XML file for Cisco Unified Wireless phone 7926G, perform the following steps:

Prerequisites

Cisco Unified CME 8.6 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone tag*
4. **mac address** [*mac-address*]
5. **type** *phone-type*
6. **button** *button-number*
7. **ephone-template** *template tag*
8. **service [phone** *parameter name parameter value*] **|** **[xml-config append** *phone_service xml filename*]
9. **telephony-service**
10. **cnf-file perphone**
11. **create cnf-files**
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	**ephone** *phone-tag* **Example:** `Router(config)# ephone 1`	Enters ephone configuration mode.

	Command or Action	Purpose
Step 4	**mac-address** [*mac-address*] **Example:** `Router(config-ephone)# mac-address 0001.2345.6789`	Specifies the MAC address of the IP phone that is being configured.
Step 5	**type** *phone-type* **Example:** `Router(config-ephone)# type 7926`	Specifies the type of phone that is being configured.
Step 6	**button** *button-number* **Example:** `Router(config-ephone)# button 1:1`	Creates a set of ephone-dns overlaid on a single button.
Step 7	**ephone-template** *template tag* **Example:** `Router(config)#ephone-template 5`	Enters ephone-template configuration mode to create an ephone template.
Step 8	**service [phone** *parameter name parameter value*] **| [xml-config append** *phone_service xml filename*] **Example:** `Router(config-ephone-template)#service xml-config append flash:7926_phone_services.xml`	Sets parameters for all IP phones that support the configured functionality and to which this template is applied. • *parameter name*—The parameter name is word and case-sensitive. See *Cisco Unified CME Command Reference* for a list of parameters. • *phone_service xml filename*—Allows the addition of a phone services xml file.
Step 9	**telephony-service** **Example:** `Router(config)telephony-service`	Enters telephony-service configuration mode.
Step 10	**cnf-file perphone** **Example:** `(config-telephony)# cnf-file perphone`	Specifies that the system generates a separate configuration XML file for each IP phone. • Separate configuration files for each endpoint are required for security.
Step 11	**create cnf-files** **Example:** `Router(config-telephony)# create cnf-files`	Builds XML configuration files required for SCCP phones.
Step 12	**end** **Example:** `Router(config-telephony)#end`	Returns to privileged EXEC mode.

How to Configure Phones to Make Basic Call

- Configuring a Mixed Shared Line, page 296 (optional)
- SCCP: Configuring the Maximum Number of Calls, page 300
- SIP: Configuring the Busy Trigger Limit, page 303
- SIP: Configuring KEMs, page 305
- SIP: Provisioning Using the Fast-Track Configuration Approach, page 306

Configuring a Mixed Shared Line

To configure a mixed shared line between Cisco Unified SIP IP and Cisco Unified SCCP IP phones, perform the following steps.

Prerequisites

Cisco Unified CME 9.0 or a later version.

Restrictions

- Cisco Unified SCCP trunk-dn is not supported.
- Mixed shared lines can only be configured on one of several common directory numbers.
- Mixed shared lines are not supported in Cisco Unified SRST.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number** *number*
5. **shared-line** [**max calls** *number-of-calls*]
6. **exit**
7. **ephone-dn** *dn-tag* [**dual-line** | **octo-line**]
8. **number** *number*
9. **shared-line sip**
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	**voice register dn** *dn-tag* **Example:** `Router(config)# voice register dn 1`	Enters voice register dn configuration mode. • *dn-tag*—Unique sequence number that identifies a particular directory number during configuration tasks. Range is 1 to 150 or the maximum defined by the **max-dn** command.
Step 4	**number** *number* **Example:** `Router(config-register-dn)# number 1001`	Associates a telephone or extension number with a Cisco Unified SIP IP phone in a Cisco Unified CME system. • *number*—String of up to 16 characters that represents an E.164 telephone number. Normally, the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number.
Step 5	**shared-line** [**max-calls** *number-of-calls*] **Example:** `Router(config-register-dn)# shared-line max-calls 4`	Creates a directory number to be shared by multiple Cisco Unified SIP IP phones. • **max-calls** *number-of-calls*—(Optional) Maximum number of active calls allowed on the shared line. Range: 2 to 16. Default: 2.
Step 6	**exit** **Example:** `Router(config-register-dn)# exit`	Exits voice register dn configuration mode.
Step 7	**ephone-dn** *dn-tag* [**dual-line** | **octo-line**] **Example:** `Router(config)# ephone-dn 1 octo-line`	Enters ephone-dn configuration mode to configure a directory number for an IP phone line. • *dn-tag*—Unique number that identifies an ephone-dn during configuration tasks. Range is 1 to the number set by the **max-dn** command. • **dual-line**—(Optional) Enables two calls per directory number. • **octo-line**—(Optional) Enables eight calls per directory number.

	Command or Action	Purpose
Step 8	**number** *number* **Example:** `Router(config-ephone-dn)# number 1001`	Associates a telephone or extension number with this 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 may contain alphabetic characters when the number is dialed only by the router, as with an intercom number.
Step 9	**shared-line sip** **Example:** `Router(config-ephone-dn)# shared-line sip`	Adds an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP and Cisco Unified SCCP IP phones.
Step 10	**end** **Example:** `Router(config-ephone-dn)# end`	Exits to privileged EXEC mode.

Troubleshooting Tips

Use the **debug ephone shared-line-mixed** command to display debugging information about mixed shared lines.

The following is a sample output from the **debug ephone shared-line-mixed** command for an outgoing call:

```
Router# debug ephone shared-line-mixed
Mar  9 20:16:37.571: skinny_notify_shrl_state_change: shrl event 1 sccp_id 0 peer_tag
20014 callid 53 incoming 0
Mar  9 20:16:37.571: skinny_shrl_get_call_state: dn 14, chan 1 call state 0
Mar  9 20:16:37.571: skinny_shrl_reserve_idle_chan: reserve dn 14, chan 1
Mar  9 20:16:37.571: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 1
Mar  9 20:16:37.583: skinny_process_shrl_event: event type 1 callid 53 dn 14 chan 1

Mar  9 20:16:37.583: skinny_process_shrl_callproc: dn 14, chan 1, callid 53
Mar  9 20:16:37.583: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Router#
Router#
Mar  9 20:16:45.151: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag
20014 callid 53 incoming 0
Mar  9 20:16:45.151: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar  9 20:16:45.155: skinny_process_shrl_event: event type 2 callid 53 dn 14 chan 1

Mar  9 20:16:45.155: skinny_update_shrl_remote: incoming 0, remote_number 2509,
remote_name 2509
Router#
Router#
Mar  9 20:16:57.775: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag
20014 callid 53 incoming 0
Mar  9 20:16:57.779: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar  9 20:16:57.779: skinny_process_shrl_event: event type 4 callid 53 dn 14 chan 1

Mar  9 20:16:57.779: skinny_update_shrl_call_state: dn 14, chan 1, call state 2
```

The following is a sample output from the **debug ephone shared-line-mixed** command for an incoming call with Hold and Resume:

```
Router# debug ephone shared-line-mixed
Mar  9 20:17:16.943: skinny_update_shrl_dn_chan: dn 14, chan 1
Mar  9 20:17:19.143: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag
20014 callid 57 incoming 1
Mar  9 20:17:19.143: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar  9 20:17:19.147: skinny_process_shrl_event: event type 2 callid 57 dn 14 chan 1

Mar  9 20:17:19.147: skinny_update_shrl_remote: incoming 1, remote_number 2509,
remote_name 2509
Mar  9 20:17:19.155: skinny_shrl_get_call_state: dn 14, chan 1 call state 2
Mar  9 20:17:19.155: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar  9 20:17:19.159: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1

Mar  9 20:17:19.159: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Router#
Mar  9 20:17:24.347: skinny_notify_shrl_state_change: shrl event 4 sccp_id 112 peer_tag
20014 callid 57 incoming 0
Mar  9 20:17:24.347: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 4
Mar  9 20:17:24.347: skinny_process_shrl_event: event type 5 callid 57 dn 14 chan 1

Mar  9 20:17:24.347: skinny_update_shrl_call_state: dn 14, chan 1, call state 8
Mar  9 20:17:28.307: skinny_shrl_resume_non_active_line: ref 5 line 4
Mar  9 20:17:28.307: skinny_update_shrl_call_state: dn 14, chan 1, call state 2
Mar  9 20:17:28.319: skinny_shrl_resume_non_active_line: fake redial to 2509
Mar  9 20:17:29.127: skinny_shrl_check_remote_resume: resume callid 62 holder callid 57
Mar  9 20:17:29.127: skinny_shrl_check_remote_resume: resume callid 62 holder callid 57
Mar  9 20:17:29.127: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Mar  9 20:17:29.135: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag
20014 callid 57 incoming 0
Mar  9 20:17:29.135: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar  9 20:17:29.135: skinny_shrl_set_resume_info: dn 14, chan 1
Mar  9 20:17:29.135: skinny_update_shrl_dn_chan: dn 14, chan 1
Mar  9 20:17:29.155: skinny_process_shrl_event: event type 4 callid 57 dn 14 chan 1

Router
Mar  9 20:17:42.407: skinny_notify_shrl_hold_or_resume_request: dn 14, chan 1, hold 1
Mar  9 20:17:42.411: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Router#
Mar  9 20:17:46.979: skinny_notify_shrl_state_change: shrl event 1 sccp_id 112 peer_tag
20014 callid 64 incoming 0
Mar  9 20:17:46.979: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 1
Mar  9 20:17:46.983: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Mar  9 20:17:46.987: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag
20014 callid 64 incoming 0
Mar  9 20:17:46.987: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar  9 20:17:46.987: skinny_process_shrl_event: event type 1 callid 64 dn 14 chan 1

Mar  9 20:17:46.987: skinny_process_shrl_event: event type 2 callid 64 dn 14 chan 1

Mar  9 20:17:46.999: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar  9 20:17:46.999: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar  9 20:17:47.007: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1

Mar  9 20:17:47.007: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Mar  9 20:17:47.007: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
```

```
Router#
Mar  9 20:17:53.795: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag
20014 callid 64 incoming 0
Mar  9 20:17:53.795: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar  9 20:17:53.795: skinny_process_shrl_event: event type 4 callid 64 dn 14 chan 1

Mar  9 20:17:53.795: skinny_update_shrl_call_state: dn 14, chan 1, call state 2
```

SCCP: Configuring the Maximum Number of Calls

To configure the maximum number of calls on a Cisco Unified SCCP IP phone in Cisco Unified CME 9.0, perform the following steps.

Prerequisites

- Cisco Unified CME 9.0 and later versions.
- Correct firmware, 9.2(1) or a later version, is installed.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag* [**dual-line** | **octo-line**]
4. **number** *number*
5. **exit**
6. **ephone** *phone-tag*
7. **mac-address** *mac-address*
8. **type** *phone-type*
9. **busy-trigger-per-button** *number-of-calls*
10. **max-calls-per-button** *number-of-calls*
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.

	Command or Action	Purpose
Step 3	**ephone-dn** *dn-tag* [**dual-line** | **octo-line**] **Example:** `Router(config)# ephone-dn 6 octo-line`	Enters ephone-dn configuration mode to configure a directory number for an IP phone line. • *dn-tag*—Unique number that identifies an ephone-dn during configuration tasks. Range is 1 to the number set by the **max-dn** command. • **dual-line**—(Optional) Enables two calls per directory number. • **octo-line**—(Optional) Enables eight calls per directory number.
Step 4	**number** *number* **Example:** `Router(config-ephone-dn)# number 1007`	Associates a telephone or extension number with an ephone-dn in a Cisco Unified CME. • *number*—String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number. One or more periods (.) can be used as wildcard characters.
Step 5	**exit** **Example:** `Router(config-ephone-dn)# exit`	Exits ephone-dn configuration mode.
Step 6	**ephone** *phone-tag* **Example:** `Router(config)# ephone 98`	Enters ephone configuration mode. • *phone-tag*—Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific.Type ? to display range.
Step 7	**mac-address** *mac-address* **Example:** `Router(config-ephone)# mac-address ABCD.1234.56EF`	Associates the MAC address of a Cisco IP phone with an ephone configuration in a Cisco Unified CME. • *mac-address*—Identifying MAC address of an IP phone.
Step 8	**type** *phone-type* **Example:** `Router(config-ephone)# type 8941`	Assigns a phone type to an SCCP phone.
Step 9	**busy-trigger-per-button** *number-of-calls* **Example:** `Router(config-ephone)# busy-trigger-per-button 6`	Sets the maximum number of calls allowed on an octo-line directory number before activating Call Forward Busy or a busy tone. • *number-of-calls*—Maximum number of calls. Range: 1 to 8. Default: 0 (disabled).

	Command or Action	Purpose
Step 10	**max-calls-per-button** *number-of-calls* **Example:** `Router(config-ephone)# max-calls-per-button 4`	Sets the maximum number of calls allowed on an octo-line directory number on an SCCP phone. • *number-of-calls*—Maximum number of calls. Range: 1 to 8. Default: 8.
Step 11	**end** **Example:** `Router(config-ephone)# end`	Exits configuration mode and enters privileged EXEC mode.

SIP: Configuring the Busy Trigger Limit

To configure the busy trigger limit on a Cisco Unified SIP IP phone in Cisco Unified CME 9.0, perform the following steps.

Prerequisites

- Cisco Unified CME 9.0 and later versions.
- Correct firmware is installed:
  - 9.2(1) or a later version for Cisco Unified 6921, 6941, 6945 and 6961 SIP IP phones.
  - 9.2(2) or a later version for Cisco Unified 8941 and 8945 SIP IP phones.

Restrictions

You cannot configure the maximum number of calls per line. The phone controls the maximum number of outgoing calls.

Table 26 shows the maximum number of outgoing calls allowed by a phone and the maximum number of incoming calls that can be configured using the **busy-trigger-per-button** command for Cisco Unified 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP Phones in Cisco Unified CME 9.0.

Table 26 Maximum Number of Incoming and Outgoing Calls

Cisco Unified SIP IP Phones	Maximum Number of Outgoing Calls (Controlled by Phones)	Maximum Number of Incoming Calls Before Busy Tone (Configurable)
6921	12	12
6941	24	24
6945	24	24
6961	72	72
8941	24	24
8945	24	24

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **type** *phone-type*
5. **busy-trigger-per-button** *number*
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 20`	Enters voice register pool configuration mode and creates a pool configuration for a SIP IP phone in Cisco Unified CME. *pool-tag*—Unique number assigned to the pool. Range is 1 to 100. **Note** For Cisco Unified CME systems, the upper limit for this argument is defined by the **max-pool** command.
Step 4	**`type`** *`phone-type`* **Example:** `Router(config-register-pool)# type 6921`	Defines a phone type for a SIP phone.
Step 5	**`busy-trigger-per-button`** *`number`* **Example:** `Router(config-register-pool)# busy-trigger-per-button 25`	Sets the maximum number of calls allowed on a SIP directory number before activating Call Forward Busy or a busy tone. • *number*—Maximum number of calls. Range: 1 to the maximum number of incoming calls listed in Step 6. The default values are 1 for the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP phones and 2 for the Cisco Unified 8941 and 8945 SIP IP phones.
Step 6	**`end`** **Example:** `Router(config-register-pool)# end`	Exits configuration mode and enters privileged EXEC mode.

SIP: Configuring KEMs

To configure KEMs for Cisco Unified 8961, 9951, or 9971 SIP IP phones, perform the following steps.

Prerequisites

Cisco Unified CME 9.1 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **type** *phone-type* [**addon 1 CKEM** [**2 CKEM** [**3 CKEM**]]]

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 pool** *pool-tag* **Example:** `Router(config)# voice register pool 29`	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME. • *pool-tag*—Unique number assigned to the pool. Range is 1 to 100. **Note** For Cisco Unified CME systems, the upper limit for this argument is defined by the **max-pool** command.
Step 4	**type** *phone-type* [**addon 1 CKEM** [**2 CKEM** [**3 CKEM**]]] **Example:** `Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM`	Defines a phone type for a Cisco Unified SIP IP phone. The following keywords increase the number of speed-dial, busy-lamp-field, and directory number keys that can be configured: • **addon 1 CKEM**—(Optional) Tells the router that a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP Phone. **Note** This option is available to Cisco Unified 8961, 9951, and 9971 SIP IP phones only. • **2 CKEM**—(Optional) Tells the router that a second Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP Phone. **Note** This option is available to Cisco Unified 9951 and 9971 SIP IP phones only. • **3 CKEM**—(Optional) Tells the router that a third Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP Phone. **Note** This option is available to Cisco Unified 9971 SIP IP phones only.

SIP: Provisioning Using the Fast-Track Configuration Approach

To provision the Cisco Unified SIP IP phones using the fast-track configuration approach, perform the following steps.

Prerequisites

You require Cisco Unified CME Release 10 or a later release.

Restrictions

When a new Cisco Unified SIP IP phone is configured on Cisco Unified CME using the fast-track configuration approach, and the Cisco Unified CME is upgraded to a later version that supports the new phone type, the fast-track configuration pertaining to that SIP IP phone is removed automatically.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool-type** *pool-type*
4. **addons** *max-addon count*
5. **description** *string*
6. **gsm-support**
7. **num-lines** *number*
8. **phoneload-support**
9. **reference-pooltype** *pool-type*
10. **telnet-support**
11. **transport** *transport-type*
12. **xml-config** *xml-tag value*
13. **exit**
14. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	`enable` **Example:** `Router> enable`	Enables the privileged EXEC mode. Enter your password if prompted.
Step 2	`configure terminal` **Example:** `Router# configure terminal`	Enters the global configuration mode.
Step 3	`voice register pool-type` **Example:** `Router(config)# voice register pool-type 9900`	Enters the voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME. If the new phone type is an existing phone that is supported on Cisco Unified CME release, you get the following error message: `ERROR: 8945 is built-in phonemodel, cannot be changed`
Step 4	`addons` *max-addons* **Example:** `Router(config-register-pooltype)# addons 3`	Defines the maximum number of add-on modules supported in Cisco Unified SIP IP phones. • *max-addons*—The maximum allowed value is 3. The configured add-on modules can be used while defining the pool for the new SIP phone model using the existing **type** command as shown below: **type** <**phone-type**> [addon 1 module-type [2 module-type]]

	Command or Action	Purpose
Step 5	**description** *string* **Example:** `Router(config-register-pooltype)# description TEST PHON`	Defines the description string for the new phone type.
Step 6	**gsm-support** **Example:** `Router(config-register-pooltype)# gsm-support`	Defines phone support for Global System for Mobile Communications (GSM) support.
Step 7	**num-lines** *max-lines* **Example:** `Router(config-register-pooltype)# num-lines 12`	Defines the maximum number of lines supported by the new phone. • *max-lines*—If this parameter is not configured, the default value 1 is used.
Step 8	**Phoneload-support** **Example:** `Router(config-register-pooltype)# Phoneload-support`	Defines phone support for firmware download from Cisco Unified CME. You can use the **load** command in the voice register global mode to configure the corresponding phone load for the new phone type if it supports phone load.
Step 9	**telnet-support** **Example:** `Router(config-register-pooltype)# telnet-support`	Defines phone support for Telnet access.
Step 10	**transport** {**udp** | **TCP**} **Example:** `Router(config-register-pooltype)# transport TCp`	Defines the default transport type supported by the new phone. If this parameter is not configured, UDP is used as the default value. The **session-transport** command configured at the voice register pool takes priority over this configuration.
Step 11	**Xml-config {maxNumCalls | busyTrigger | custom}** **Example:** `Router(config-register-pooltype)#xml-config busyTrigger 2` `Router(config-register-pooltype)#xml-config maxNumCalls 4` `Router(config-register-pooltype)#xml-config custom <test>1</test>`	Defines the phone-specific XML tags to be used in the configuration file. • **maxNumCalls**— Defines the maximum number of calls allowed per line. • **busyTrigger**— Defines the number of calls that triggers Call Forward Busy per line on the SIP phone. • **custom**—Defines custom XML tags which can be appended at the end of the phone specific CNF file. These parameters are used while generating the configuration profile file.

	Command or Action	Purpose
Step 12	**exit** **Example:** `Router(config-register-pooltype)# exit`	Exits the voice register-pooltype configuration mode.
Step 13	**end** **Example:** `Router(config)# end`	Exits the privileged EXEC configuration mode.

Configuration Examples for Making Basic Calls

This section contains the following examples of the required Cisco Unified CME configurations with some of the additional options that are discussed in other modules.

- Configuring SCCP Phones for Making Basic Calls: Example, page 310
- Configuring SIP Phones for Making Basic Calls: Example, page 314
- Disabling a Bulk Registration for a SIP Phone: Example, page 317
- Configuring a Mixed Shared Line on a Second Common Directory Number: Example, page 317
- Cisco ATA: Example, page 318
- SCCP Analog Phone: Example, page 318
- Remote Teleworker Phones: Example, page 319
- Secure IP Phone (IP-STE): Example, page 319
- Configuring PhoneServices XML File for Cisco Unified Wireless Phone 7926G: Example, page 320
- Monitoring the Status of Key Expansion Modules: Example, page 320
- Example: Fast-Track Configuration Approach, page 322

Configuring SCCP Phones for Making Basic Calls: Example

The following is a sample output of the **show running-config** command, showing how an SCCP phone is configured to make basic calls:

```
Router# show running-config

version 12.4
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CME40
!
boot-start-marker
boot-end-marker
!
logging buffered 2000000 debugging
!
no aaa new-model
!
resource policy
!
clock timezone PST -8
clock summer-time PDT recurring
no network-clock-participate slot 2
voice-card 0
 no dspfarm
 dsp services dspfarm
!
voice-card 2
 dspfarm
!
no ip source-route
```

```
ip cef
!
!
!
ip domain name cisco.com
ip multicast-routing
!
!
ftp-server enable
ftp-server topdir flash:
isdn switch-type primary-5ess
!
!
!
voice service voip
 allow-connections h323 to sip
 allow-connections sip to h323
 no supplementary-service h450.2
 no supplementary-service h450.3
 h323
  call start slow
!
!
!
controller T1 2/0/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 2/0/1
 framing esf
 linecode b8zs
!
!
interface GigabitEthernet0/0
 ip address 192.168.1.1 255.255.255.0
 ip pim dense-mode
 duplex auto
 speed auto
 media-type rj45
 negotiation auto
!
interface Service-Engine1/0
 ip unnumbered GigabitEthernet0/0
 service-module ip address 192.168.1.2 255.255.255.0
 service-module ip default-gateway 192.168.1.1
!
interface Serial2/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 isdn map address ^.* plan unknown type international
 no cdp enable
!
!
ip route 0.0.0.0 0.0.0.0 192.168.1.254
ip route 192.168.1.2 255.255.255.255 Service-Engine1/0
ip route 192.168.2.253 255.255.255.255 10.2.0.1
ip route 192.168.3.254 255.255.255.255 10.2.0.1
!
!
ip http server
ip http authentication local
```

```
no ip http secure-server
ip http path flash:
!
!
!
!
tftp-server flash:P00307020300.loads
tftp-server flash:P00307020300.sb2
tftp-server flash:P00307020300.sbn
!
control-plane
!
!
!
voice-port 2/0/0:23
!
!
!
sccp local GigabitEthernet0/0
sccp ccm 192.168.1.1 identifier 1
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register MTP0013c49a0cd0
 keepalive retries 5
!
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec gsmfr
 codec g729r8
 maximum sessions 90
 associate application SCCP
!
!
dial-peer voice 9000 voip
 mailbox-selection last-redirect-num
 destination-pattern 78..
 session protocol sipv2
 session target ipv4:192.168.1.2
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
dial-peer voice 2 pots
 incoming called-number .
 direct-inward-dial
 port 2/0/0:23
 forward-digits all
!
dial-peer voice 1 pots
 destination-pattern 9[2-9]......
 port 2/0/0:23
 forward-digits 8
!
dial-peer voice 3 pots
 destination-pattern 91[2-9]..[2-9]......
 port 2/0/0:23
 forward-digits 12!
!
gateway
```

```
 timer receive-rtp 1200
!
!
telephony-service
 load 7960-7940 P00307020300
 max-ephones 100
 max-dn 300
 ip source-address 192.168.1.1 port 2000
 system message CCME 4.0
 sdspfarm units 1
 sdspfarm transcode sessions 128
 sdspfarm tag 1 MTP0013c49a0cd0
 voicemail 7800
 max-conferences 24 gain -6
 call-forward pattern .T
 moh music-on-hold.au
 multicast moh 239.1.1.1 port 2000
 web admin system name admin password sjdfg
 transfer-system full-consult
 transfer-pattern .T
 secondary-dialtone 9
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
!
ephone-dn-template  1
!
!
ephone-template  1
 keep-conference endcall local-only
 codec g729r8 dspfarm-assist
!
!
ephone-template  2
!
!
ephone-dn  1
 number 6001
 call-forward busy 7800
 call-forward noan 7800 timeout 10
!
!
ephone-dn  2
 number 6002
 call-forward busy 7800
 call-forward noan 7800 timeout 10
!
!
ephone-dn  10
 number 6013
 paging ip 239.1.1.1 port 2000
!
!
ephone-dn  20
 number 8000....
 mwi on
!
!
ephone-dn  21
 number 8001....
 mwi off
!
!
!
!
```

```
ephone  1
 device-security-mode none
 username "user1"
 mac-address 002D.264E.54FA
 codec g729r8 dspfarm-assist
 type 7970
 button  1:1
!
!
!
ephone  2
 device-security-mode none
 username "user2"
 mac-address 001C.821C.ED23
 type 7960
 button  1:2
!
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line 66
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output all
line 258
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output all
line vty 0 4
 exec-timeout 0 0
 privilege level 15
 password sgpxw
 login
!
scheduler allocate 20000 1000
ntp server 192.168.224.18
!
!
end
```

Configuring SIP Phones for Making Basic Calls: Example

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
 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
!
 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
!
voice register pool 5
 id mac 0012.019B.3FD8
 type ATA
 number 1 dn 5
 preference 1
 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
 dtmf-relay-rtp-nte
 call-forward b2bua all 7778
!
voice register pool 7
!
voice register pool 8
 id mac 0006.D737.CC42
 type 7940
 number 1 dn 8
 template 2
 preference 1
 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
```

Disabling a Bulk Registration for a SIP Phone: Example

The following example shows that all phone numbers that match the pattern “408555..” can register with the SIP proxy server (IP address 1.5.49.240) except directory number 1, number “4085550101,” for which bulk registration is disabled:

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

Configuring a Mixed Shared Line on a Second Common Directory Number: Example

The following example shows how configuring a mixed shared line on a second common directory number is rejected:

```
Router(config)#ephone-dn 14 octo-line
Router(config-ephone-dn)#number 2502
Router(config-ephone-dn)#shared-line sip

Router(config)#ephone-dn 20 octo-line
Router(config-ephone-dn)#number 2502
Router(config-ephone-dn)#shared-line sip
DN number already exists in the shared line database
```

Cisco ATA: Example

The following example shows the configuration for two analog phones using a single Cisco ATA with MAC address 000F.F758.E70E. The analog phone attached to the first port uses the MAC address of the Cisco ATA. The analog phone attached to the second port uses a modified version of the Cisco ATA's MAC address; the first two hexadecimal numbers are removed and 01 is appended to the end.

```
telephony-service
 conference hardware
 load ATA ATA030203SCCP051201A.zup
!
ephone-dn  80  dual-line
 number 8080
!
ephone-dn  81  dual-line
 number 8081
!
ephone  30
 mac-address 000F.F758.E70E
 type ata
 button  1:80
!
ephone  31
 mac-address 0FF7.58E7.0E01
 type ata
 button  1:81
```

SCCP Analog Phone: Example

The following partial sample output from a Cisco Unified CME configuration sets transfer type to full-blind and sets the voice-mail extension to 5200. Ephone-dn 10 has the extension 4443 and is assigned to Tommy; that number and name will be used for caller-ID displays. The description field under ephone-dn is used to indicate that this ephone-dn is on the Cisco VG224 voice gateway at port 1/3. Extension 4443 is assigned to ephone 7, which is an analog phone type with 10 speed-dial numbers.

```
CME_Router# show running-config
.
.
.
telephony-service
 load 7910 P00403020214
 load 7960-7940 P00305000301
 load 7905 CP79050101SCCP030530B31
 max-ephones 60
 max-dn 60
 ip source-address 10.8.1.2 port 2000
 auto assign 1 to 60
 create cnf-files version-stamp 7960 Sep 28 2004 17:23:02
 voicemail 5200
 mwi relay
 mwi expires 99999
 max-conferences 8 gain -6
 web admin system name cisco password lab
 web admin customer name ac2 password cisco
 dn-webedit
 time-webedit
 transfer-system full-blind
 transfer-pattern 6...
 transfer-pattern 5...
!
```

```
!
ephone-dn  10  dual-line
 number 4443 secondary 9191114443
 pickup-group 5
 description vg224-1/3
 name tommy
!
ephone  7
 mac-address C863.9018.0402
 speed-dial 1 4445
 speed-dial 2 4445
 speed-dial 3 4442
 speed-dial 4 4441
 speed-dial 5 6666
 speed-dial 6 1111
 speed-dial 7 1112
 speed-dial 8 9191114441
 speed-dial 9 9191114442
 speed-dial 10 9191114442
 type anl
 button  1:10
```

Remote Teleworker Phones: Example

The following example shows the configuration for ephone 270, a remote teleworker phone with its codec set to G.729r8. The **dspfarm-assist** keyword is used to ensure that calls from this phone will use DSP resources to maintain the G.729r8 codec when calls would normally be switched to a G.711 codec.

```
ephone 270
 button 1:36
 mtp
 codec g729r8 dspfarm-assist
 description teleworker remote phone
```

Secure IP Phone (IP-STE): Example

The following example shows the configuration for Secure IP Phone IP-STE. IP-STE is the phone type required to configure a secure phone.

```
ephone-dn  1
 number 3001
...
ephone  9
 mac-address 0004.E2B9.1AD1
 max-calls-per-button 1
 type IP-STE
 button  1:1 2:2 3:3 4:4
```

Configuring PhoneServices XML File for Cisco Unified Wireless Phone 7926G: Example

The following example shows phone type 7926 configured in ephone 1 and service xml-config file configured in ephone template 1:

```
!
!
!
telephony-service
 max-ephones 58
 max-dn 192
 ip source-address 1.4.206.105 port 2000
 cnf-file perphone
 create cnf-files
!
ephone-template 1
 service xml-config append flash:7926_phone_services.xml
!
ephone-dn  1 octo-line
  number 1001
!
ephone  1
  mac-address AAAA.BBBB.CCCC
  ephone-template 1
  type 7926
  button 1:1
!
```

Monitoring the Status of Key Expansion Modules: Example

Show commands are used to monitor the status and other details of Key Expansion Modules (KEMs).

The following example demonstrates how the **show voice register all** command displays KEM details with all the Cisco Unified CME configurations and registration information:

```
show voice register all
VOICE REGISTER GLOBAL
=====================
CONFIG [Version=9.1]
========================
...........
Pool Tag 5
Config:
  Mac address is B4A4.E328.4698
  Type is 9971 addon 1 CKEM
  Number list 1 : DN 2
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Video is enabled
  Camera is enabled
  Busy trigger per button value is 0
  keep-conference is enabled
  registration expires timer max is 200 and min is 60
  kpml signal is enabled
  Lpcor Type is none
```

The following example demonstrates how the **show voice register pool type** command displays all the phones configured with add-on KEMs in Cisco Unified CME:

```
Router# show voice register pool type CKEM
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
4    B4A4.E328.4698  9.45.31.111     1  4   5589$                REGISTERED
```

Example: Fast-Track Configuration Approach

The following example shows how to enable the new Cisco Unified 9900 SIP IP phone to inherit the properties of the Cisco Unified SIP IP phone 9951 and overwrite some of the phone's properties:

```
voice register pool-type 9900
    reference-pooltype  9951
    description  SIP Phone 9900 addon module
    num-lines  24
    addons 3
    no phoneload-support
    xml-config custom "custom-sftp"1"/custom-sftp"

voice register pool 1
    type 9900 addon 1 CKEM 2 CKEM 3 CKEM
    id mac 1234.4567.7891
voice register global
    mode cme
    load 9900 P0S3-06-0-00
```

The following example shows how to inherit the existing properties of a reference phone type (Cisco Unified SIP IP phone 6921) using the fast-track configuration approach.

```
voice register pooltype 6922
    reference-pooltype 6921
    device-name  "SIP Phone 6922"

voice register pool 11
    type 6922
    id mac 1234.4567.7890
```

Where to Go Next

To select a fixed-button layout for a Cisco Unified IP Phone 7931G, see the "SCCP: Selecting Button Layout for a Cisco Unified IP Phone 7931G" section on page 1476.

After configuring phones in Cisco Unified CME to make basic calls, you are ready to generate configuration files for the phones to be connected to your router. See the "Generating Configuration Files for Phones" section on page 359.

Additional References

The following sections provide references related to Cisco Unified CME features.

Related Documents

Related Topic	Document Title
Cisco Unified CME configuration	• *Cisco Unified CME Command Reference* • Cisco Unified CME Documentation Roadmap
Cisco IOS commands	• *Cisco IOS Voice Command Reference* • Cisco IOS Software Releases 12.4T command references
Cisco IOS configuration	• *Cisco IOS Voice Configuration Library* • Cisco IOS Software Releases 12.4T Configuration Guides
Phone documentation for Cisco Unified CME	• User Documentation for Cisco Unified IP Phones

Technical Assistance

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

Feature Information for Configuring Phones to Make Basic Calls

Caution The Interactive Voice Response (IVR) media prompts feature is only available on the IAD2435 when running IOS version 15.0(1)M or later.

Table 27 lists the features in this module and enhancements to the features by version.

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

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS 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.

Note Table 27 lists the Cisco Unified CME version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified CME software also support that feature.

Table 27 Feature Information for Basic Call Features

Feature Name	Cisco Unified CME Versions	Feature Information
KEM Support for Cisco Unified 8961, 9951, and 9971 SIP IP Phones	9.1	Increases line key and feature key appearances, speed dials, or programmable buttons on Cisco Unified SIP IP phones.
Cisco ATA-187	9.0	Supports T.38 fax relay and fax pass-through on Cisco ATA-187.
Cisco Unified SIP IP Phones		Adds SIP support for the following phone types: • Cisco Unified 6901 and 6911 IP Phones • Cisco Unified 6921, 6941, 6945, and 6961 IP Phones • Cisco Unified 8941 and 8945 IP Phones
Mixed Shared Lines		Allows Cisco Unified SIP and SCCP IP phones to share a common directory number.
Multiple Calls Per Line		Overcomes the limitation on the maximum number of calls per line.
Real-Time Transport Protocol Call Information Display Enhancement	8.8	Allows you to display information on active RTP calls using the **show ephone rtp connections** command. The output from this command provides an overview of all the connections in the system, narrowing the criteria for debugging pulse code modulation and Cisco Unified CME packets without a sniffer.
Support for Cisco Unified 3905 SIP IP Phones		Adds support for SIP phones connected to a Cisco Unified CME system.
Support for Cisco Unified 6945, 8941, and 8945 SCCP IP Phones		Adds support for SCCP phones connected to a Cisco Unified CME system.

Table 27 Feature Information for Basic Call Features (continued)

Feature Name	Cisco Unified CME Versions	Feature Information
Support for 7926G Wireless SCCP IP Phone	8.6	Added support for 7926G Wireless SCCP IP Phone.
Secure IP Phones	8.0	Adds support for Secure IP Phone (IP-STE).
SIP Shared Lines	7.1	Adds support for nonexclusive shared lines on SIP phones.
Autoconfiguration for Cisco VG202, VG204, and VG224		Adds autoconfiguration for the Cisco VG202, VG204, and VG224 Analog Phone Gateway.
Ephone-Type Templates	7.0/4.3	Adds support for dynamically adding new phone types without upgrading Cisco IOS software.
Octo-Line Directory Numbers		Adds octo-line directory numbers that support up to eight active calls.
G.722 and iLBC Transcoding and Conferencing Support in Cisco Unified CME		Adds support for the G.722-64K and iLBC codecs.
Dial Plans for SIP Phones	4.1	Adds support for dial plans for SIP phones.
KPML		Adds support for KPML for SIP phones.
Session Transport Protocol		Adds selection for session-transport protocol for SIP phones.
Watch Mode		Provides Busy Lamp Field (BLF) notification on a line button that is configured for watch mode on one phone for all lines on another phone (watched phone) for which the watched directory number is the primary line.
Remote Teleworker Phones	4.0	Introduces support for teleworker remote phones.
Analog Phones	4.0	Introduces support for analog phones with SCCP supplementary features using FXS ports on Cisco Integrated Services Routers.
	3.2.1	Introduces support for analog phones with SCCP supplementary features using FXS ports on a Cisco VG224 voice gateway.
	3.0	Introduces support for Cisco ATA 186 and Cisco ATA 188.
	1.0	Introduces support for analog phones in H.323 mode using FXS ports.
Cisco IP Communicator	4.0	Introduces support for Cisco IP Communicator.

Table 27 Feature Information for Basic Call Features (continued)

Feature Name	Cisco Unified CME Versions	Feature Information
Direct FXO Trunk Lines	4.0	Adds enhancements to improve the keyswitch emulation behavior of PSTN lines in a Cisco Unified CME system, including the following: • Status monitoring of the FXO port on the line button of an IP phone. • Transfer recall if a transfer-to phone does not answer after a specified timeout. • Transfer-to button optimization to free up the private extension line on the transfer-to phone • Directory numbers for FXO lines can be configured for dual-line to support the FXO monitoring, transfer recall, and transfer-to button optimization features.
	3.2	Introduces direct FXO trunk line capability.
SIP Phones	3.4	Adds support for SIP phones connected to Cisco CME system.
Monitor Mode for Shared Lines	3.0	Provides a visible line status indicating whether the line is in-use or not.