Voice Mail Integration

This chapter describes how to integrate your voice-mail system with Cisco Unified Communications Manager Express (Cisco Unified CME).

Prerequisites for Voice Mail Integration

- Calls can be successfully completed between phones on the same Cisco Unified CME router.
- If your voice-mail system is something other than Cisco Unity Express, such as Cisco Unity, voice mail must be installed and configured on your network.
- If your voice-mail system is Cisco Unity Express:

Note When you order Cisco Unity Express, Cisco Unity Express software and the purchased license are installed on the module at the factory. Spare modules also ship with the software and license installed. If you are adding Cisco Unity Express to an existing Cisco router, you will be required to install hardware and software components.

  - Interface module for Cisco Unity Express is installed. For information about the AIM-CUE or NM-CUE, access documents located at http://www.cisco.com/en/US/products/hw/modules/ps2797/prod_installation_guides_list.html.
  - The recommended Cisco IOS release and feature set plus the necessary Cisco Unified CME phone firmware files to support Cisco Unity Express are installed on the Cisco Unified CME router.

    To determine whether the Cisco IOS software release and Cisco Unified CME software version are compatible with the Cisco Unity Express version, Cisco router model, and Cisco Unity Express hardware that you are using, see Cisco Unity Express Compatibility Matrix.

To verify installed Cisco Unity Express software version, enter the Cisco Unity Express command environment and use the **show software version** user EXEC command. For information about the command environment, see the appropriate *Cisco Unity Express CLI Administrator Guide* at http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/roadmap/cuedocs.html.

- The proper license for Cisco Unified CME, not Cisco Unified Communications Manager, is installed. To verify installed license, enter the Cisco Unity Express command environment and use the **show software license** user EXEC command. For information about the command environment, see the appropriate *Cisco Unity Express CLI Administrator Guide* at http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/roadmap/cuedocs.html.

  This is an example of the Cisco Unified CME license:

```
se-10-0-0-0> show software licenses
Core:
- application mode: CCME
- total usable system ports: 8

Voicemail/Auto Attendant:
- max system mailbox capacity time: 6000
- max general delivery mailboxes: 15
- max personal mailboxes: 50

Languages:
- max installed languages: 1
- max enabled languages: 1
```

- Voicemail and Auto Attendant (AA) applications are configured. For configuration information, see *"Configuring the System Using the Initialization Wizard"* in the appropriate Cisco Unity Express GUI Administrator Guide at http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/roadmap/cuedocs.html.

Information About Voice-Mail Integration

Cisco Unity Connection Integration

Cisco Unity Connection transparently integrates messaging and voice recognition components with your data network to provide continuous global access to calls and messages. These advanced, convergence-based communication services help you use voice commands to place calls or listen to messages in "hands-free" mode and check voice messages from your desktop, either integrated into an e-mail inbox or from a Web browser. Cisco Unity Connection also features robust automated-attendant functions that include intelligent routing and easily customizable call-screening and message-notification options.

For instructions on how to integrate Cisco Unified CME with Cisco Unity Connection, see Cisco CallManager Express 3.x Integration Guide for Cisco Unity Connection 1.1.

Cisco Unity Express Integration

Cisco Unity Express offers easy, one-touch access to messages and commonly used voice-mail features that enable users to reply, forward, and save messages. To improve message management, users can create alternate greetings, access envelope information, and mark or play messages based on privacy or urgency. For instructions on how to configure Cisco Unity Express, see the administrator guides for Cisco Unity Express.

For configuration information, see Enable DTMF Integration Using SIP NOTIFY.

Note Cisco Unified CME and Cisco Unity Express must both be configured before they can be integrated.

Cisco Unity Integration

Cisco Unity is a Microsoft Windows-based communications solution that brings you voice mail and unified messaging and integrates them with the desktop applications you use daily. Cisco Unity gives you the ability to access all of your messages, voice, fax, and e-mail, by using your desktop PC, a touchtone phone, or the Internet. The Cisco Unity voice mail system supports voice-mail integration with Cisco Unified CME. This integration requires that you configure the Cisco Unified CME router and Cisco Unity software to get voice-mail service.

For configuration instructions, see Enable DTMF Integration Using RFC 2833.

DTMF Integration for Legacy Voice-Mail Applications

For dual-tone multifrequency (DTMF) integrations, information on how to route incoming or forwarded calls is sent by a telephone system in the form of DTMF digits. The DTMF digits are sent in a pattern that is based on the integration file in the voice-mail system connected to the Cisco Unified CME router. These patterns are required for DTMF integration of Cisco Unified CME with most voice-mail systems. Voice-mail systems are designed to respond to DTMF after the system answers the incoming calls.

After configuring the DTMF integration patterns on the Cisco Unified CME router, you set up the integration files on the third-party legacy voice-mail system by following the instructions in the documents that accompany the voice-mail system. You must design the DTMF integration patterns appropriately so that the voice-mail system and the Cisco Unified CME router work with each other.

For configuration information, see Enable DTMF Integration for Analog Voice-Mail Applications.

Mailbox Selection Policy

Typically a voice-mail system uses the number that a caller has dialed to determine the mailbox to which a call should be sent. However, if a call has been diverted several times before reaching the voice-mail system, the mailbox that is selected might vary for different types of voice-mail systems. For example, Cisco Unity Express uses the last number to which the call was diverted before it was sent to voice mail as the mailbox number. Cisco Unity and some legacy PBX systems use the originally called number as the mailbox number.

The Mailbox Selection Policy feature allows you to provision the following options from the Cisco Unified CME configuration.

- For Cisco Unity Express, you can select the originally dialed number.
- For PBX voice-mail systems, you can select the last number to which the call was diverted before it was sent to voice mail. This option is configured on the outgoing dial peer for the voice-mail system's pilot number.

- For Cisco Unity voice mail, you can select the last number to which the call was diverted before it was sent to voice mail. This option is configured on the ephone-dn that is associated with the voice-mail pilot number.

To enable Mailbox Selection Policy, see Set a Mailbox Selection Policy for Cisco Unity Express or a PBX Voice-Mail Number or Set a Mailbox Selection Policy for Cisco Unity.

RFC 2833 DTMF MTP Pass through

In Cisco Unified CME 4.1, the RFC 2833 Dual-Tone Multifrequency (DTMF) Media Termination Point (MTP) Passthrough feature provides the capability to pass DTMF tones transparently between SIP endpoints that require transcoding or Resource Reservation Protocol (RSVP) agents.

This feature supports DTMF Relay across SIP WAN devices that support RFC 2833, such as Cisco Unity and SIP trunks. Devices registered to a Cisco Unified CME SIP back-to-back user agent (B2BUA) can exchange RFC 2833 DTMF MTP with other devices that are not registered with the Cisco Unified CME SIP B2BUA, or with devices that are registered in one of the following:

- Local or remote Cisco Unified CME
- Cisco Unified Communications Manager
- Third party proxy

By default, the RFC 2833 DTMF MTP Passthrough feature uses payload type 101 on MTP, and MTP accepts all the other dynamic payload types if it is indicated by Cisco Unified CME. For configuration information, see Enable DTMF Integration Using RFC 2833.

MWI Line Selection

Message waiting indicator (MWI) line selection allows you to choose the phone line that is monitored for voice-mail messages and that lights an indicator when messages are present.

Before Cisco Unified CME 4.0, the MWI lamp on a phone running SCCP could be associated only with the primary line of the phone.

In Cisco Unified CME 4.0 and later versions, you can designate a phone line other than the primary line to be associated with the MWI lamp. Lines other than the one associated with the MWI lamp display an envelope icon when a message is waiting. A logical phone “line” is not the same as a phone button. A button with one or more directory numbers is considered one line. A button with no directory number assigned does not count as a line.

In Cisco Unified CME 4.0 and later versions, a SIP directory number that is used for call forward all, presence BLF status, and MWI features must be configured by using the **dn** keyword in the **number** command; direct line numbers are not supported.

For configuration information, see Configure a Voice Mailbox Pilot Number on a SCCP Phone or Configure a Directory Number for MWI NOTIFY.

AMWI

The AMWI (Audible Message Line Indicator) feature provides a special stutter dial tone to indicate message waiting. This is an accessibility feature for vision-impaired phone users. The stutter dial tone is defined as 10 ms ON, 100 ms OFF, repeat 10 times, then steady on.

In Cisco Unified CME 4.0(3), you can configure the AMWI feature on the Cisco Unified IP Phone 7911 and Cisco Unified IP Phone 7931G to receive audible, visual, or audible and visual MWI notification from an external voice-messaging system. AMWI cannot be enabled unless the **number** command is already configured for the IP phone to be configured.

Cisco Unified CME applies the following logic based on the capabilities of the IP phone and how MWI is configured:

- If the phone supports (visual) MWI and MWI is configured for the phone, activate the Message Waiting light.
- If the phone supports (visual) MWI only, activate the Message Waiting light regardless of the configuration.
- If the phone supports AMWI and AMWI is configured for the phone, send the stutter dial tone to the phone when it goes off-hook.
- If the phone supports AMWI only and AMWI is configured, send the stutter dial tone to the phone when it goes off-hook regardless of the configuration.

If a phone supports (visual) MWI and AMWI and both options are configured for the phone, activate the Message Waiting light and send the stutter dial tone to the phone when it goes off-hook.

For configuration information, see Configure a SCCP Phone for MWI Outcall.

SIP MWI Prefix Specification

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco Unified CME 4.0 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for MWI that include a prefix string as a site identifier.

For example, an MWI message might indicate that the central mailbox number 555-0123 has a voice message. In this example, the digits 555 are set as the prefix string or site identifier using the **mwi prefix** command. The local Cisco Unified CME system is able to convert 555-0123 to 0123 and deliver the MWI to the correct phone. Without this prefix string manipulation, the system would reject an MWI for 555-0123 as not matching the local Cisco Unified CME extension 0123.

To enable SIP MWI Prefix Specification, see Enable SIP MWI Prefix Specification.

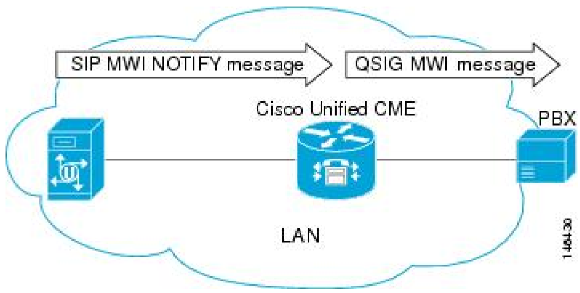
SIP MWI - QSIG Translation

In Cisco Unified CME 4.1 and later, the SIP MWI - QSIG Translation feature extends MWI functionality for SIP MWI and QSIG MWI interoperation to enable sending and receiving MWI over QSIG to a PBX.

When the SIP Unsolicited NOTIFY is received from voice mail, the Cisco router translates this event to activate QSIG MWI to the PBX, via PSTN. The PBX will switch on, or off, the MWI lamp on the corresponding IP phone. This feature supports only Unsolicited NOTIFY. Subscribe NOTIFY is not supported by this feature.

In Figure 1: SIP MWI to ISDN QSIG When Voice Mail and Cisco Router are On the Same LAN, on page 6, the Cisco router receives the SIP Unsolicited NOTIFY, performs the protocol translation, and initiates the QSIG MWI call to the PBX, where it is routed to the appropriate phone.

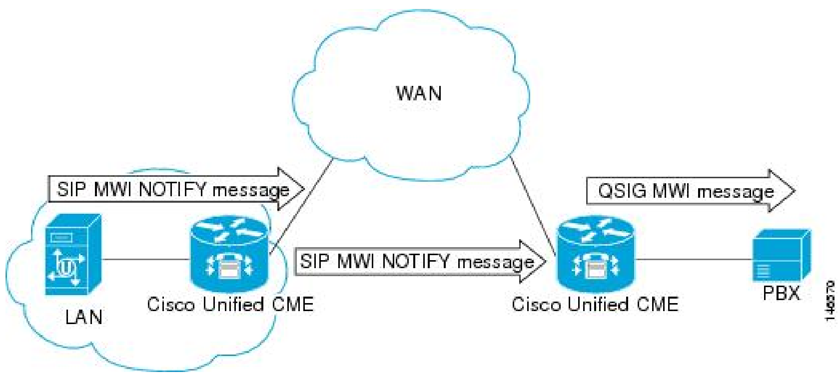
Figure 1: SIP MWI to ISDN QSIG When Voice Mail and Cisco Router are On the Same LAN



It makes no difference if the SIP Unsolicited NOTIFY is received via LAN or WAN if the PBX is connected to the Cisco router, and not to the remote voice-mail server.

In Figure 2: SIP MWI to ISDN QSIG When PBX is Connected to a Remote Cisco Router, on page 6, a voice mail server and Cisco Unified CME are connected to the same LAN and a remote Cisco Unified CME is connected across the WAN. In this scenario, the protocol translation is performed at the remote Cisco router and the QSIG MWI message is sent to the PBX.

Figure 2: SIP MWI to ISDN QSIG When PBX is Connected to a Remote Cisco Router



VMWI

There are two types of visual message waiting indicator (VMWI) features: Frequency-shift Keying (FSK) and DC voltage. The message-waiting lamp can be enabled to flash on an analog phone that requires an FSK message to activate a visual indicator. The DC Voltage VMWI feature is used to flash the message-waiting lamp on an analog phone which requires DC voltage instead of an FSK message. For all other applications, such as MGCP, FSK VMWI is used even if the voice gateway is configured for DC voltage VMWI. The configuration for DC voltage VMWI is supported only for Foreign Exchange Station (FXS) ports on the Cisco VG224 analog voice gateway with analog device version V1.3 and V2.1.

The Cisco VG224 can only support 12 Ringer Equivalency Number (REN) for ringing 24 onboard analog FXS voice ports. To support ringing and DC Voltage VMWI for 24 analog voice ports, stagger-ringing logic is used to maximize the limited REN resource. When a system runs out of REN because too many voice ports are being rung, the MWI lamp temporarily turns off to free up REN to ring the voice ports.

DC voltage VMWI is also temporarily turned off any time the port's operational state is no longer idle and onhook, such as when one of the following events occur:

- Incoming call on voice port
- Phone goes off hook
- The voice port is shut down or busied out

Once the operational state of the port changes to idle and onhook again, the MWI lamp resumes flashing until the application receives a requests to clear it; for example, if there are no more waiting messages.

For configuration information, see Transfer to Voice Mail.

Transfer to Voice Mail

The Transfer to Voice Mail feature allows a phone user to transfer a caller directly to a voice-mail extension. The user presses the TrnsfVM softkey to place the call on hold, enters the extension number, and then commits the transfer by pressing the TrnsfVM softkey again. The caller hears the complete voice mail greeting. This feature is supported using the TrnsfVM softkey or feature access code (FAC).

For example, a receptionist might screen calls for five managers. If a call comes in for a manager who is not available, the receptionist can transfer the caller to the manager's voice-mail extension by using the TrnsfVM softkey and the caller hears the personal greeting of the individual manager.

For configuration information, see Transfer to Voice Mail.

Live Record

The Live Record feature enables IP phone users in a Cisco Unified CME system to record a phone conversation if Cisco Unity Express is the voice mail system. An audible notification, either by announcement or by periodic beep, alerts participants that the conversation is being recorded. The playing of the announcement or beep is under the control of Cisco Unity Express.

Live Record is supported for two-party calls and ad hoc conferences. In normal record mode, the conversation is recorded after the LiveRcd softkey is pressed. This puts the other party on-hold and initiates a call to Cisco Unity Express at the configured live-record number. To stop the recording session, the phone user presses the LiveRcd softkey again, which toggles between on and off.

The Live-Record number is configured globally and must match the number configured in Cisco Unity Express. You can control the availability of the feature on individual phones by modifying the display of the LiveRcd softkey using an ephone template. This feature must be enabled on both Cisco Unified CME and Cisco Unity Express.

To enable Live Record in Cisco Unified CME, see Configure Live Record on SCCP Phones.

Cisco Unity Express AXL Enhancement

In Cisco Unified CME 7.0(1) and later versions, the Cisco Unity Express AXL enhancement in Cisco Unified CME provides better administrative integration between Cisco Unified CME and Cisco Unity Express by automatically synchronizing passwords.

No configuration is required to enable this feature.

Configure Voice-Mail Integration

Configure a Voice Mailbox Pilot Number on a SCCP Phone

To configure the telephone number that is speed-dialed when the Message button on a SCCP phone is pressed, perform the following steps.

Note The same telephone number is configured for voice messaging for all SCCP phones in Cisco Unified CME.

Before you begin

- Voicemail phone number must be a valid number; directory number and number for voicemail phone number must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **voicemail** *phone-number*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters voice register global configuration mode to set parameters for all supported phones in Cisco Unified CME.
Step 4	**voicemail** *phone-number* **Example:** `Router(config-telephony)# voice mail 0123`	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed. • *phone-number*—Same phone number is configured for voice messaging for all SCCP phones in a Cisco Unified CME.

	Command or Action	Purpose
Step 5	**end** **Example:** `Router(config-telephony)# end`	Exits to privileged EXEC mode.

What to do next

- (Cisco Unified CME 4.0 or a later version only) To set up a mailbox selection policy, see Configure a Mailbox Selection Policy on SCCP Phone.
- To set up DTMF integration patterns for connecting to analog voice-mail applications, see Enable DTMF Integration for Analog Voice-Mail Applications.
- To connect to a remote SIP-based IVR or Cisco Unity, or to connect to a remote SIP-PSTN that goes through the PSTN to a voice-mail or IVR application, see Enable DTMF Integration Using RFC 2833.
- To connect to a Cisco Unity Express system, configure a nonstandard SIP NOTIFY format. See Enable DTMF Integration Using SIP NOTIFY.

Configure a Mailbox Selection Policy on SCCP Phone

Perform *one* of the following tasks, depending on which voice-mail application is used:

- Set a Mailbox Selection Policy for Cisco Unity Express or a PBX Voice-Mail Number
- Set a Mailbox Selection Policy for Cisco Unity

Set a Mailbox Selection Policy for Cisco Unity Express or a PBX Voice-Mail Number

To set a policy for selecting a mailbox for calls from a Cisco Unified CME system that are diverted before being sent to a Cisco Unity Express or PBX voice-mail pilot number, perform the following steps.

Restriction In the following scenarios, the mailbox selection policy can fail to work properly:

- The last redirecting endpoint is not hosted on Cisco Unified CME. This may rarely occur with a PBX.
- A call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- A call is forwarded across non-Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

Before you begin

Cisco Unified CME 4.0 or a later version.

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **dial-peer voice** *tag* **voip** or **dial-peer voice** *tag* **pots**
4. **mailbox-selection** [**last-redirect-num** | **orig-called-num**]
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	**dial-peer voice** *tag* **voip** or **dial-peer voice** *tag* **pots**
	Example:
	`Router(config)# dial-peer voice 7000 voip`
	or
	`Router(config)# dial-peer voice 35 pots`
		Enters dial-peer configuration mode.
		• *tag*—identifies the dial peer. Valid entries are 1 to 2147483647.
		Note Use this command on the outbound dial peer associated with the pilot number of the voice-mail system. For systems using Cisco Unity Express, this is a VoIP dial peer. For systems using PBX-based voice mail, this is a POTS dial peer.

Step 4	**mailbox-selection** [**last-redirect-num** | **orig-called-num**] **Example:** `Router(config-dial-peer)# mailbox-selection orig-called-num`	Sets a policy for selecting a mailbox for calls that are diverted before being sent to a voice-mail line. • **last-redirect-num**—(PBX voice mail only) The mailbox number to which the call will be sent is the last number to divert the call (the number that sends the call to the voice-mail pilot number). • **orig-called-num**—(Cisco Unity Express only) The mailbox number to which the call will be sent is the number that was originally dialed before the call was diverted.
Step 5	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

What to do next

- To use voice mail on a SIP network that connects to a Cisco Unity Express system, configure a nonstandard SIP NOTIFY format. See Enable DTMF Integration Using SIP NOTIFY.

Set a Mailbox Selection Policy for Cisco Unity

To set a policy for selecting a mailbox for calls that are diverted before being sent to a Cisco Unity voice-mail pilot number, perform the following steps.

Restriction This feature might not work properly in certain network topologies, including when:

- The last redirecting endpoint is not hosted on Cisco Unified CME. This may rarely occur with a PBX.
- A call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- A call is forwarded across other voice gateways that do not support the optional H450.3 originalCalledNr field.

Before you begin

- Cisco Unified CME 4.0 or a later version.
- Directory number to be configured is associated with a voice mailbox.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **exit**
4. **ephone-dn** *dn-tag*
5. **mailbox-selection** [**last-redirect-num**]
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	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 4	**ephone-dn** *dn-tag* **Example:**	Enters ephone-dn configuration mode.

	Command or Action	Purpose
	`Router(config)# ephone-dn 752`	
Step 5	**mailbox-selection** [**last-redirect-num**] **Example:** `Router(config-ephone-dn)# mailbox-selection last-redirect-num`	Sets a policy for selecting a mailbox for calls that are diverted before being sent to a Cisco Unity voice-mail pilot number.
Step 6	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

What to do next

- To use a remote SIP-based IVR or Cisco Unity, or to connect Cisco Unified CME to a remote SIP-PSTN that goes through the PSTN to a voice-mail or IVR application, see Enable DTMF Integration Using RFC 2833.

Transfer to Voice Mail

To enable a phone user to transfer a call to voice mail by using the TrnsfVM softkey or a FAC, perform the following steps.

Restriction The TrnsfVM softkey is not supported on the Cisco Unified IP Phone 7905, 7912, or 7921, or analog phones connected to the Cisco VG224 or Cisco ATA. These phones support the trnsfvm FAC.

Before you begin

- Cisco Unified CME 4.3 or a later version.
- Cisco Unity Express 3.0 or a later version, installed and configured.
- For information about standard and custom FACs, see Configuring Feature Access Codes.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-template** *template-tag*
4. **softkeys connected** { [**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**] }
5. **exit**
6. **ephone** *phone-tag*
7. **ephone-template** *template-tag*
8. **exit**
9. **telephony-service**

10. **voicemail** *phone-number*
11. **fac** {**standard** | **custom trnsfvm** *custom-fac*}
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-template** *template-tag* **Example:** `Router(config)# ephone-template 5`	Enters ephone-template configuration mode to create an ephone template. • *template-tag*—Unique identifier for the ephone template. Range: 1 to 20.
Step 4	**softkeys connected** {[**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**]} **Example:** `Router(config-ephone-template)# softkeys connected TrnsfVM Park Acct ConfList Confrn Endcall Trnsfer Hold`	(Optional) Modifies the order and type of softkeys that display on an IP phone during the connected call state. • You can enter any of the keywords in any order. • Default is all softkeys are displayed in alphabetical order. • Any softkey that is not explicitly defined is disabled.
Step 5	**exit** **Example:** `Router(config-ephone-template)# exit`	Exits ephone-template configuration mode.
Step 6	**ephone** *phone-tag* **Example:** `Router(config)# ephone 12`	Enters ephone configuration mode. • *phone-tag*—Unique number that identifies this ephone during configuration tasks.
Step 7	**ephone-template** *template-tag* **Example:** `Router(config-ephone)# ephone-template 5`	Applies the ephone template to the phone. • *template-tag*—Unique identifier of the ephone template that you created in Step 3, on page 13.
Step 8	**exit** **Example:** `Router(config-ephone)# exit`	Exits ephone configuration mode.

	Command or Action	Purpose
Step 9	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 10	**voicemail** *phone-number* **Example:** `Router(config-telephony)# voicemail 8900`	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed. • *phone-number*—Same phone number is configured for voice messaging for all SCCP phones in a Cisco Unified CME.
Step 11	**fac** {**standard** | **custom trnsfvm** *custom-fac*} **Example:** `Router(config-telephony)# fac custom trnsfvm #22`	Enables standard FACs or creates a custom FAC or alias. • **standard**—Enables standard FACs for all phones. Standard FAC for transfer to voice mail is *6. • **custom**—Creates a custom FAC for a FAC type. • *custom-fac*—User-defined code to be dialed using the keypad on an IP or analog phone. Custom FAC can be up to 256 characters long and contain numbers 0 to 9 and * and #.
Step 12	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Example

The following example shows a configuration where the display order of the TrnsfVM softkey is modified for the connected call state in ephone template 5 and assigned to ephone 12. A custom FAC for transfer to voice mail is set to #22.

```
telephony-service
 max-ephones 100
 max-dn 240
 timeouts transfer-recall 60
 voicemail 8900
 max-conferences 8 gain -6
 transfer-system full-consult
 fac custom trnsfvm #22
!
!
ephone-template  5
 softkeys connected  TrnsfVM Park Acct ConfList Confrn Endcall Trnsfer Hold
 max-calls-per-button 3
 busy-trigger-per-button 2
!
!
ephone  12
 ephone-template 5
 mac-address 000F.9054.31BD
```

```
type 7960
button  1:10 2:7
```

What to do next

- If you are finished modifying parameters for phones in Cisco Unified CME, generate a new configuration file and restart the phones. See SCCP: Generating Configuration Files for SCCP Phones.
- For information on how phone users transfer a call to voice mail, see Cisco Unified IP Phone documentation for Cisco Unified CME.

Configure Live Record on SCCP Phones

To configure the Live Record feature so that a phone user can record a conversation by pressing the LiveRcd softkey, perform the followings steps.

Restriction

- Only one live record session is allowed for each conference.
- Only the conference creator can initiate a live record session. In an ad hoc conference, participants who are not the conference creator cannot start a live record session. In a two-party call, the party who starts the live record session is the conference creator.

Note For legal disclaimer information about this feature, see copyright information section.

Before you begin

- Cisco Unified CME 4.3 or a later version.
- Cisco Unity Express 3.0 or a later version, installed and configured. For information on configuring Live Record in Cisco Unity Express, see Configure Live Record in the *Cisco Unity Express Voice-Mail and Auto-Attendant CLI Administrator Guide for 3.0 and Later Versions*.
- Ad hoc hardware conference resource is configured and ready to use. See Configure Hardware Conferencing.
- If phone user wants to view the live record session, include ConfList softkey using the **softkeys** connected command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **live record** *number*
5. **voicemail** *number*
6. **exit**

7. **ephone-dn** *dn-tag*
8. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
9. **call-forward all** *target-number*
10. **exit**
11. **ephone-template** *template-tag*
12. **softkeys connected** {[**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**]}
13. **exit**
14. **ephone** *phone-tag*
15. **ephone-template** *template-tag*
16. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**live record** *number* **Example:** `Router(config-telephony)# live record 8900`	Defines the extension number that is dialed when the LiveRcd softkey is pressed on an SCCP IP phone.
Step 5	**voicemail** *number* **Example:** `Router(config-telephony)# voicemail 8000`	Defines the extension number that is speed-dialed when the Messages button is pressed on an IP phone. • *Number*—Cisco Unity Express voice-mail pilot number.
Step 6	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 7	**ephone-dn** *dn-tag* **Example:** `Router(config)# ephone-dn 10`	Creates a directory number that forwards all calls to the Cisco Unity Express voice-mail pilot number.
Step 8	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]	Assigns an extension number to this directory number.

	Command or Action	Purpose
	Example: `Router(config-ephone-dn)# number 8900`	• *Number*—Must match the Live Record pilot-number configured in Step 4, on page 16.
Step 9	**call-forward all** *target-number* **Example:** `Router(config-ephone-dn)# call-forward all 8000`	Forwards all calls to this extension to the specified voice-mail number. • *target-number*—Phone number to which calls are forwarded. Must match the voice-mail pilot number configured in Step 5, on page 16. **Note** Phone users can activate and cancel the call-forward-all state from the phone using the CFwdAll softkey or a FAC.
Step 10	**exit** **Example:** `Router(config-ephone-dn)# exit`	Exits ephone-dn configuration mode.
Step 11	**ephone-template** *template-tag* **Example:** `Router(config)# ephone-template 5`	Enters ephone-template configuration mode to create an ephone template. • *template-tag*—Unique identifier for the ephone template. Range: 1 to 20.
Step 12	**softkeys connected** { [**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**] } **Example:** `Router(config-ephone-template)# softkeys connected LiveRcd Confrn Hold Park Trnsfer TrnsfVM`	Modifies the order and type of softkeys that display on an IP phone during the connected call state.
Step 13	**exit** **Example:** `Router(config-ephone-template)# exit`	Exits ephone-template configuration mode.
Step 14	**ephone** *phone-tag* **Example:** `Router(config)# ephone 12`	Enters ephone configuration mode. • *phone-tag*—Unique number that identifies this ephone during configuration tasks.
Step 15	**ephone-template** *template-tag* **Example:** `Router(config-ephone)# ephone-template 5`	Applies the ephone template to the phone. • *template-tag*—Unique identifier of the ephone template that you created in Step 11, on page 17.
Step 16	**end** **Example:**	Exits to privileged EXEC mode.

	Command or Action	Purpose
	`Router(config-ephone)# end`	

Example

The following example shows Live Record is enabled at the system-level for extension 8900. All incoming calls to extension 8900 are forwarded to the voice-mail pilot number 8000 when the LiveRcd softkey is pressed, as configured under ephone-dn 10. Ephone template 5 modifies the display order of the LiveRcd softkey on IP phones.

```
telephony-service
 privacy-on-hold
 max-ephones 100
 max-dn 240
 timeouts transfer-recall 60
 live-record 8900
 voicemail 8000
 max-conferences 8 gain -6
 transfer-system full-consult
 fac standard
!
!
ephone-template  5
 softkeys remote-in-use  CBarge Newcall
 softkeys hold  Resume Newcall Join
 softkeys connected  LiveRcd Confrn Hold Park Trnsfer TrnsfVM
 max-calls-per-button 3
 busy-trigger-per-button 2
!
!
ephone-dn  10
 number 8900
 call-forward all 8000
```

Configure a Voice Mailbox Pilot Number on a SIP Phone

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

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

Before you begin

- Directory number and number for voicemail phone number must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice register global**
4. **voicemail** *phone-number*
5. **exit**
6. **voice register dn** *dn-tag*
7. **call-forward b2bua busy** *directory-number*
8. **call-forward b2bua mailbox** *directory-number*
9. **call-forward b2bua noan** *directory-number* **timeout** *seconds*
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 global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**voicemail** *phone-number* **Example:** `Router(config-register-global)# voice mail 1111`	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed. • *phone-number*—Same phone number is configured for voice messaging for all SIP phones in a Cisco Unified CME.
Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 6	**voice register dn** *dn-tag* **Example:** `Router(config)# voice register dn 2`	Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI.
Step 7	**call-forward b2bua busy** *directory-number* **Example:** `Router(config-register-dn)# call-forward b2bua busy 1000`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that is busy will be forwarded to the designated directory number.
Step 8	**call-forward b2bua mailbox** *directory-number* **Example:**	Designates the voice mailbox to use at the end of a chain of call forwards.

	Command or Action	Purpose
	`Router(config-register-dn)# call-forward b2bua mailbox 2200`	• Incoming calls have been forwarded to a busy or no-answer extension will be forwarded to the directory-number specified.
Step 9	**call-forward b2bua noan** *directory-number* **timeout** *seconds* **Example:** `Router(config-register-dn)# call-forward b2bua noan 2201 timeout 15`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that does not answer will be forwarded to the designated directory number. • *seconds*—Number of seconds that a call can ring with no answer before the call is forwarded to another extension. Range: 3 to 60000. Default: 20.
Step 10	**end** **Example:** `Router(config-register-dn)# end`	Exits to privileged EXEC mode.

What to do next

- To set up DTMF integration patterns for connecting to analog voice-mail applications, see Enable DTMF Integration for Analog Voice-Mail Applications.
- To use a remote SIP-based IVR or Cisco Unity, or to connect to a remote SIP-PSTN that goes through the PSTN to a voice-mail or IVR application, see Enable DTMF Integration Using RFC 2833.
- To connect to a Cisco Unity Express system, configure a nonstandard SIP NOTIFY format, see Enable DTMF Integration Using SIP NOTIFY.

Enable DTMF Integration

Perform *one* of the following tasks, depending on which DTMF-relay method is required:

- Enable DTMF Integration for Analog Voice-Mail Applications—To set up DTMF integration patterns for connecting to analog voice-mail applications.
- Enable DTMF Integration Using RFC 2833—To connect to a remote SIP-based IVR or voice-mail application such as Cisco Unity or when SIP is used to connect Cisco Unified CME to a remote SIP-PSTN voice gateway that goes through the PSTN to a voice-mail or IVR application.
- Enable DTMF Integration Using SIP NOTIFY—To configure a SIP dial peer to point to Cisco Unity Express.

Enable DTMF Integration for Analog Voice-Mail Applications

To set up DTMF integration patterns for analog voice-mail applications, perform the following steps.

Note You can configure multiple tags and tokens for each pattern, depending on the voice-mail system and type of access.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **vm-integration**
4. **pattern direct** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
5. **pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
6. **pattern ext-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
7. **pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
8. **pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-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	**vm-integration** **Example:** `Router(config) vm-integration`	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail system.
Step 4	**pattern direct** *tag1* {**CGN**	**CDN**	**FDN**} [*tag2* {**CGN**	**CDN**	**FDN**}] [*tag3* {**CGN**	**CDN**	**FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration) pattern direct 2 CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone. • The *tag* attribute is an alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number. • The keywords, **CGN**, **CDN**, and **FDN**, configure the type of call information sent to the voice-mail system, such as calling number (CGN), called number (CDN), or forwarding number (FDN).

	Command or Action	Purpose
Step 5	**pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration) pattern ext-to-ext busy 7 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail.
Step 6	**pattern ext-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration) pattern ext-to-ext no-answer 5 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an internal extension fails to connect to an extension and the call is forwarded to voice mail.
Step 7	**pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches a busy extension and the call is forwarded to voice mail.
Step 8	**pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration)# pattern trunk-to-ext no-answer 4 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
Step 9	**end** **Example:** `Router(config-vm-integration)# exit`	Exits configuration mode and enters privileged EXEC mode.

What to do next

After configuring DTMF relay, you are ready to configure Message Waiting Indicator (MWI) notification for either the MWI outcall, unsolicited notify, or subscribe/notify mechanism. See Configure a SCCP Phone for MWI Outcall.

Enable DTMF Integration Using RFC 2833

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

This DTMF relay method is required in the following situations:

- When SIP is used to connect Cisco Unified CME to a remote SIP-based IVR or voice-mail application such as Cisco Unity.

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

Note If the T.38 Fax Relay feature is also configured on this IP network, we recommend that you either configure the voice gateways to use a payload type other than PT96 or PT97 for fax relay negotiation, or depending on whether the SIP endpoints support different payload types, configure Cisco Unified CME to use a payload type other than PT96 or PT97 for DTMF.

Before you begin

- Configure the **codec** or **voice-class codec** command for transcoding between G.711 and G.729.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **description** *string*
5. **destination-pattern** *string*
6. **session protocol sipv2**
7. **session target** {**dns:** *address* | **ipv4:** *destination-address*}
8. **dtmf-relay rtp-nte**
9. **dtmf-interworking rtp-nte**
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	**dial-peer voice** *tag* **voip** **Example:** `Router (config)# dial-peer voice 123 voip`	Enters dial-peer configuration mode to define a VoIP dial peer for the voice-mail system. • *tag*—Defines the dial peer being configured. Range is 1 to 2147483647.
Step 4	**description** *string* **Example:**	(Optional) Associates a description with the dial peer being configured. Enter a string of up to 64 characters.

	Command or Action	Purpose
	`Router (config-voice-dial-peer)# description CU pilot`	
Step 5	**destination-pattern** *string* **Example:** `Router (config-voice-dial-peer)# destination-pattern 20`	Specifies the pattern of the numbers that the user must dial to place a call. • *string*—Prefix or full E.164 number.
Step 6	**session protocol sipv2** **Example:** `Router (config-voice-dial-peer)# session protocol sipv2`	Specifies that Internet Engineering Task Force (IETF) Session Initiation Protocol (SIP) is protocol to be used for calls between local and remote routers using the packet network.
Step 7	**session target** {**dns :** *address* | **ipv4 :** *destination-address*} **Example:** `Router (config-voice-dial-peer)# session target ipv4:10.8.17.42`	Designates a network-specific address to receive calls from the dial peer being configured. • **dns :** *address*—Specifies the DNS address of the voice-mail system. • **ipv4 :** *destination- address*—Specifies the IP address of the voice-mail system.
Step 8	**dtmf-relay rtp-nte** **Example:** `Router (config-voice-dial-peer)# dtmf-relay rtp-nte`	Sets DTMF relay method for the voice dial peer being configured. • **rtp-nte**— Provides conversion from the out-of-band SCCP indication to the SIP standard for DTMF relay (RFC 2833). Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type. • This command can also be configured in voice-register-pool configuration mode. For individual phones, the phone-level configuration for this command overrides the system-level configuration for this command. **Note** The need to use out-of-band conversion is limited to SCCP phones. SIP phones natively support in-band.
Step 9	**dtmf-interworking rtp-nte** **Example:** `Router (config-voice-dial-peer)# dtmf-interworking rtp-nte`	(Optional) Enables a delay between the dtmf-digit begin and dtmf-digit end events in the RFC 2833 packets. • This command is supported in Cisco IOS Release 12.4(15)XZ and later releases and in Cisco Unified CME 4.3 and later versions. • This command can also be configured in voice-service configuration mode.

	Command or Action	Purpose
Step 10	**end** **Example:** `Router(config-voice-dial-peer)# end`	Exits to privileged EXEC mode.

What to do next

After configuring DTMF relay, you are ready to configure Message Waiting Indicator (MWI) notification for either the MWI outcall, unsolicited notify, or subscribe/notify mechanism. See Configure a SCCP Phone for MWI Outcall.

Enable DTMF Integration Using SIP NOTIFY

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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **description** *string*
5. **destination-pattern** *string*
6. **b2bua**
7. **session protocol sipv2**
8. **session target** {**dns** **:** *address* | **ipv4** **:** *destination-address*}
9. **dtmf-relay sip-notify**
10. **codec** *g711ulaw*
11. **no vad**
12. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal#`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router (config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode to define a VoIP dial peer for the voice-mail system. • *tag*—Defines the dial peer being configured. Range is 1 to 2147483647.

	Command or Action	Purpose
Step 4	**description** *string* **Example:** `Router (config-voice-dial-peer)# description cue pilot`	(Optional) Associates a description with the dial peer being configured. Enter a string of up to 64 characters.
Step 5	**destination-pattern** *string* **Example:** `Router (config-voice-dial-peer)# destination-pattern 20`	Specifies the pattern of the numbers that the user must dial to place a call. • *string*—Prefix or full E.164 number.
Step 6	**b2bua** **Example:** `Router (config-voice-dial-peer)# b2bua`	(Optional) Includes the Cisco Unified CME address as part of contact in 3XX response to point to Cisco Unity Express and enables SIP-to-SCCP call forward.
Step 7	**session protocol sipv2** **Example:** `Router (config-voice-dial-peer)# session protocol sipv2`	Specifies that Internet Engineering Task Force (IETF) Session Initiation Protocol (SIP) is protocol to be used for calls between local and remote routers using the packet network.
Step 8	**session target** {**dns :** *address* | **ipv4 :** *destination-address*} **Example:** `Router (config-voice-dial-peer)# session target ipv4:10.5.49.80`	Designates a network-specific address to receive calls from the dial peer being configured. • **dns :** *address*—Specifies the DNS address of the voice-mail system. • **ipv4 :** *destination- address*—Specifies the IP address of the voice-mail system.
Step 9	**dtmf-relay sip-notify** **Example:** `Router (config-voice-dial-peer)# dtmf-relay sip-notify`	Sets the DTMF relay method for the voice dial peer being configured. • **sip-notify**— Forwards DTMF tones using SIP NOTIFY messages. • This command can also be configured in voice-register-pool configuration mode. For individual phones, the phone-level configuration for this command overrides the system-level configuration for this command.
Step 10	**codec** *g711ulaw* **Example:** `Router (config-voice-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for a dial peer being configured.
Step 11	**no vad** **Example:** `Router (config-voice-dial-peer)# no vad`	Disables voice activity detection (VAD) for the calls using the dial peer being configured.

	Command or Action	Purpose
Step 12	**end** **Example:** `Router(config-voice-dial-peer)# end`	Exits to privileged EXEC mode.

What to do next

After configuring DTMF relay, you are ready to configure Message Waiting Indicator (MWI). See Configure a SCCP Phone for MWI Outcall.

Configure a SCCP Phone for MWI Outcall

To designate a phone line or directory number on an individual SCCP phone to be monitored for voice-mail messages, or to enable audible MWI, perform the following steps.

Restriction

- Audible MWI is supported only in Cisco Unified CME 4.0(2) and later versions.
- Audible MWI is supported only on Cisco Unified IP Phone 7931G and Cisco Unified IP Phone 7911.

Before you begin

- Directory number and number for MWI line must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone-tag*
4. **mwi-line** *line-number*
5. **exit**
6. **ephone-dn** *dn-tag*
7. **mwi** {**off** | **on** | **on-off**}
8. **mwi-type** {**visual** | **audio** | **both**}
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:**	Enters global configuration mode.

	Command or Action	Purpose
	`Router# configure terminal`	
Step 3	**ephone** *phone-tag* **Example:** `Router(config)# ephone 36`	Enters ephone configuration mode.
Step 4	**mwi-line** *line-number* **Example:** `Router(config-ephone)# mwi-line 3`	(Optional) Selects a phone line to receive MWI treatment. • *line-number*—Number of phone line to receive MWI notification. Range: 1 to 34. Default: 1.
Step 5	**exit** **Example:** `Router(config-ephone)# exit`	Exits ephone configuration mode.
Step 6	**ephone-dn** *dn-tag* **Example:** `Router(config)# ephone-dn 11`	Enters ephone-dn configuration mode.
Step 7	**mwi** {**off** | **on** | **on-off**} **Example:** `Router(config-ephone-dn)# mwi on-off`	(Optional) Enables a specific directory number to receive MWI notification from an external voice-messaging system. **Note** This command can also be configured in ephone-dn-template configuration mode. The value that you set in ephone-dn configuration mode has priority over the value set in ephone-dn-template mode.
Step 8	**mwi-type** {**visual** | **audio** | **both**} **Example:** `Router(config-ephone-dn)# mwi-type audible`	(Optional) Specifies which type of MWI notification to be received. **Note** This command is supported only on the Cisco Unified IP Phone 7931G and Cisco Unified IP Phone 7911. **Note** This command can also be configured in ephone-dn-template configuration mode. The value that you set in ephone-dn configuration mode has priority over the value set in ephone-dn-template mode. For configuration information, see Ephone-dn Templates.
Step 9	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

Enable MWI at the System-Level on SIP Phones

To enable a message waiting indicator (MWI) at a system-level, perform the following steps.

Before you begin

- Cisco CME 3.4 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mwi reg-e164**
5. **mwi stutter**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mwi reg-e164** **Example:** `Router(config-register-global)# mwi reg-e164`	Registers full E.164 number to the MWI server in Cisco Unified CME and enables MWI.
Step 5	**mwi stutter** **Example:** `Router(config-register-global)# mwi stutter`	Enables Cisco Unified CME router at the central site to relay MWI notification to remote SIP phones.
Step 6	**end** **Example:** `Router(config-register-global)# end`	Exits to privileged EXEC mode.

Configure a Directory Number for MWI on SIP Phones

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

- Define Pilot Call Back Number for MWI Outcall
- Configure a Directory Number for MWI NOTIFY

Define Pilot Call Back Number for MWI Outcall

To designate a phone line on an individual SIP directory number to be monitored for voice-mail messages, perform the following steps.

Restriction

- For Cisco Unified CME 4.1 and later versions, the Call Forward All, Presence, and MWI features require that SIP phones must be configured with a directory number by using the **number** command with the **dn** keyword; direct line numbers are not supported.

Before you begin

- Cisco CME 3.4 or a later version.
- Directory number and number for receiving MWI must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **mwi**
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 dn** *dn-tag* **Example:** `Router(config)# 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.

	Command or Action	Purpose
Step 4	**mwi** **Example:** `Router(config-register-dn)# mwi`	Enables a specific directory number to receive MWI notification.
Step 5	**end** **Example:** `Router(config-ephone-dn)# end`	Exits to privileged EXEC mode.

Configure a Directory Number for MWI NOTIFY

To identify the MWI server and specify a directory number for receiving MWI Subscribe/NOTIFY or MWI Unsolicited NOTIFY, follow the steps in this section.

Note We recommend using the Subscribe/NOTIFY method instead of an Unsolicited NOTIFY when possible.

Restriction

- For Cisco Unified CME 4.1 and later versions, the Call Forward All, Presence, and MWI features require that SIP phones must be configured with a directory number by using the **number** command with the **dn** keyword; direct line numbers are not supported.
- The SIP MWI - QSIG Translation feature in Cisco Unified CME 4.1 does not support Subscribe NOTIFY.
- Cisco Unified IP Phone 7960, 7940, 7905, and 7911 support only Unsolicited NOTIFY for MWI.

Before you begin

- Cisco CME 3.4 or a later version.
- For Cisco Unified CME 4.0 and later, QSIQ supplementary services must be configured on the Cisco router. For information, see Enable H.450.7 and QSIG Supplementary Services at System-Level or Enable H.450.7 and QSIG Supplementary Services on a Dial Peer.
- Directory number and number for receiving MWI must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server** {**ipv4:** *destination-address* | **dns:** *host-name*} [**unsolicited**]
5. **exit**
6. **voice register dn** *dn-tag*
7. **mwi**
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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters Session Initiation Protocol (SIP) user agent (ua) configuration mode for configuring the user agent.

Step 4	**mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*} [**unsolicited**]
	Example:
	`Router(config-sip-ua)# mwi-server ipv4:1.5.49.200`
	or
	`Router(config-sip-ua)# mwi-server dns:server.yourcompany.com unsolicited`
		Specifies voice-mail server settings on a voice gateway or UA.
		Note The **sip-server** and **mwi expires commands** under the telephony-service configuration mode have been migrated to **mwi-server** to support DNS format of the SIP server.

Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits to the next highest mode in the configuration mode hierarchy.
Step 6	**voice register dn** *dn-tag* **Example:** `Router(config)# 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 7	**mwi** **Example:** `Router(config-register-dn)# mwi`	Enables a specific directory number to receive MWI notification.
Step 8	**end** **Example:** `Router(config-register-dn)# end`	Exits to privileged EXEC mode.

Enable SIP MWI Prefix Specification

To accept unsolicited SIP Notify messages for MWI that include a prefix string as a site identifier, perform the following steps.

Before you begin

- Cisco Unified CME 4.0 or a later version.
- Directory number for receiving MWI Unsolicited NOTIFY must be configured. For information, see Configure a Directory Number for MWI NOTIFY.

SUMMARY STEPS

1. **enable**
2. **telephony-service**
3. **mwi prefix** *prefix-string*
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 3	**mwi prefix** *prefix-string* **Example:** `Router(config-telephony)# mwi prefix 555`	Specifies a string of digits that, if present before a known Cisco Unified CME extension number, are recognized as a prefix. • *prefix-string*—Digit string. The maximum prefix length is 32 digits.
Step 4	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Configure VMWI on SIP Phones

To enable a VMWI, perform the following steps.

Before you begin

- Cisco IOS Release 12.4(6)T or a later version

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port*

4. **mwi**
5. **vmwi dc-voltage** or **vmwi fsk**
6. **exit**
7. **sip-ua**
8. **mwi-server** {**ipv4:***destination-address* | **dns:***host-name*} [**unsolicited**]
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-port** *port* **Example:** `Router(config)# voice-port 2/0`	Enters voice-port configuration mode. • *port*—Syntax is platform-dependent. Type **?** to determine.
Step 4	**mwi** **Example:** `Router(config-voiceport)# mwi`	Enables MWI for a specified voice port.
Step 5	**vmwi dc-voltage** or **vmwi fsk** **Example:** `Router(config-voiceport)# vmwi dc-voltage`	(Optional) Enables DC voltage or FSK VMWI on a Cisco VG224 onboard analog FXS voice port. You do not need to perform this step for the Cisco VG202 and Cisco VG204. They support FSK only. VMWI is configured automatically when MWI is configured on the voice port. This step is required for the VG224. If an FSK phone is connected to the voice port, use the **fsk** keyword. If a DC voltage phone is connected to the voice port, use the **dc-voltage** keyword.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits to the next highest mode in the configuration mode hierarchy.
Step 7	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters Session Initiation Protocol user agent configuration mode for configuring the user agent.

	Command or Action	Purpose

Step 8	**mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*} [**unsolicited**]
	Example:
	`Router(config-sip-ua)# mwi-server ipv4:1.5.49.200`
	or
	`Router(config-sip-ua)# mwi-server dns:server.yourcompany.com unsolicited`
		Specifies voice-mail server settings on a voice gateway or user agent (ua).
		Note The **sip-server** and **mwi expires** commands under the telephony-service configuration mode have been migrated to **mwi-server** to support DNS format of the Session Initiation Protocol (SIP) server.

Step 9	**end** **Example:** `Router(config-voiceport)# end`	Exits voice-port configuration mode and returns to privileged EXEC mode.

Verify Voice-Mail Integration

- Press the **Messages** button on a local phone in Cisco Unified CME and listen for the voice mail greeting.
- Dial an unattended local phone and listen for the voice mail greeting.
- Leave a test message.
- Go to the phone that you called. Verify that the [Message] indicator is lit.
- Press the **Messages** button on this phone and retrieve the voice mail message.

Configuration Examples for Voice-Mail Integration

Example for Setting up a Mailbox Selection Policy for SCCP Phones

The following example sets a policy to select the mailbox of the originally called number when a call is diverted to a Cisco Unity Express or PBX voice-mail system with the pilot number 7000.

```
dial-peer voice 7000 voip
 destination-pattern 7000
 session target ipv4:10.3.34.211
 codec g711ulaw
 no vad
 mailbox-selection orig-called-num
```

The following example sets a policy to select the mailbox of the last number that the call was diverted to before being diverted to a Cisco Unity voice-mail system with the pilot number 8000.

```
ephone-dn 825
 number 8000
 mailbox-selection last-redirect-num
```

Example for Configuring Voice Mailbox for SIP Phones

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

```
voice register global
   voicemail 1234
  !
  voice register dn 2
   number 2200
   call-forward b2bua all 1000
   call-forward b2bua mailbox 2200
   call-forward b2bua noan 2201 timeout 15
   mwi
```

Example for Configuring DTMF Integration Using RFC 2833

The following example shows the configuration for DTMF Relay using RFC 2833:

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

Example for Configuring DTMF Integration Using SIP Notify

The following example shows the configuration for DTMF using SIP Notify:

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

Example for Configuring DTMF Integration for Legacy Voice-Mail Applications

The following example sets up DTMF integration for an analog voice-mail system.

```
vm-integration
  pattern direct 2 CGN *
  pattern ext-to-ext busy 7 FDN * CGN *
  pattern ext-to-ext no-answer 5 FDN * CGN *
  pattern trunk-to-ext busy 6 FDN * CGN *
  pattern trunk-to-ext no-answer 4 FDN * CGN *
```

Example for Enabling SCCP Phone Line for MWI

The following example enables MWI on ephone 18 for line 2 (button 2), which has overlaid ephone-dns. Only a message waiting for the first ephone-dn (2021) on this line will activate the MWI lamp. Button 4 is unused. The line numbers in this example are as follows:

- Line 1—Button 1—Extension 2020
- Line 2—Button 2—Extension 2021, 2022, 2023, 2024
- Line 3—Button 3—Extension 2021, 2022, 2023, 2024 (rollover line)

- Button 4—Unused
- Line 4—Button 5—Extension 2025

```
ephone-dn 20
   number 2020

  ephone-dn 21
  number 2021

  ephone-dn 22
  number 2022

  ephone-dn 23
   number 2023

  ephone-dn 24
   number 2024

  ephone-dn 25
   number 2025

  ephone 18
  button 1:20 2o21,22,23,24,25 3x2 5:26
   mwi-line 2
```

The following example enables MWI on ephone 17 for line 3 (extension 609). In this example, the button numbers do not match the line numbers because buttons 2 and 4 are not used. The line numbers in this example are as follows:

- Line 1—Button 1—Extension 607
- Button 2—Unused
- Line 2—Button 3—Extension 608
- Button 4—Unused
- Line 3—Button 5—Extension 609

```
ephone-dn 17
   number 607

  ephone-dn 18
   number 608

  ephone-dn 19
   number 609

  ephone 25
   button 1:17 3:18 5:19
   mwi-line 3
```

Example for Configuring SIP MWI Prefix Specification

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages for known mailbox numbers using the prefix 555.

```
sip-ua
   mwi-server 172.16.14.22 unsolicited

  telephony-service
   mwi prefix 555
```

Example for Configuring SIP Directory Number for MWI Outcall

The following example shows an MWI callback pilot number:

```
voice register dn
   number 9000….
   mwi
```

Example for Configuring SIP Directory Number for MWI Unsolicited Notify

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

```
sip-ua
   mwi-server dns:server.yourcompany.com expires 60 port 5060 transport udp unsolicited

  voice register dn 1
  number 1234
  mwi
```

Example for Configuring SIP Directory Number for MWI Subscribe/NOTIFY

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

```
sip-ua
   mwi-server ipv4:1.5.49.200

  voice register dn 1
  number 1234
  mwi
```

Feature Information for Voice-Mail Integration

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

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

Table 1: Feature Information for Voice-Mail Integration

Feature Name	Cisco Unified CME Version	Feature Information
Audible MWI	4.0(2)	Provides support for selecting audible, visual, or audible and visual Message Waiting Indicator (MWI) on supported Cisco Unified IP phones.
Cisco Unity Express AXL Enhancement	7.0(1)	Cisco Unified CME and Cisco Unity Express passwords are automatically synchronized. No configuration is required for this feature.
DTMF Integration	3.4	Added support for voice messaging systems connected via a SIP trunk or SIP user agent. The standard Subscribe/NOTIFY method is preferred over an Unsolicited NOTIFY.
	2.0	DTMF integration patterns were introduced.
Live Record	4.3	Enables IP phone users in a Cisco Unified CME system to record a phone conversation if Cisco Unity Express is the voice mail system.
Mailbox Selection Policy	4.0	Mailbox selection policy was introduced.
MWI	4.0	MWI line selection of a phone line other than the primary line on a SCCP phone was introduced.
	3.4	Voice messaging systems (including Cisco Unity) connected via a SIP trunk or SIP user agent can pass a Message Waiting Indicator (MWI) that will be received and understood by a SIP phone directly connected to Cisco Unified CME.
SIP MWI Prefix Specification	4.0	SIP MWI prefix specification was introduced.
SIP MWI - QSIG Translation	4.1	Extends message waiting indicator (MWI) functionality for SIP MWI and QSIG MWI interoperation to enable sending and receiving of MWI over QSIG to PBX.
Transfer to Voice Mail	4.3	Enables a phone user to transfer a caller directly to a voice-mail extension.