Integrating Voice Mail

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

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 Voice-Mail Integration" section on page 407.

Contents

Prerequisites

- 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/ps3115/prod_installation_guides_list.html .

– The recommended Cisco IOS release and feature set plus the necessary Cisco CME phone firmware and GUI files to support Cisco Unity Express are installed on the Cisco CME router.

If the GUI files are not installed, see the "Installing Cisco Unified CME Software" section on page 92.

To determine whether the Cisco IOS software release and Cisco CME software version are compatible with the Cisco Unity Express version, Cisco router model, and Cisco Unity Express hardware that you are using, see the 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/products/sw/voicesw/ps5520/products_documentation_roadmap09186a00803f3e19.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/products/sw/voicesw/ps5520/products_documentation_roadmap09186a00803f3e19.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/products/sw/voicesw/ps5520/products_documentation_roadmap09186a00803f3e19.html.

Information About Voice-Mail Integration

To enable voice-mail support, you should understand the following concepts:

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 the *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 the "Enabling DTMF Integration Using SIP NOTIFY" section on page 393.

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 the "Enabling DTMF Integration Using RFC 2833" section on page 390.

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 the "Enabling DTMF Integration for Analog Voice-Mail Applications" section on page 388.

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 the "SCCP: Setting a Mailbox Selection Policy for Cisco Unity Express or a PBX Voice-Mail Number" section on page 383 or the "SCCP: Setting Mailbox Selection Policy for Cisco Unity" section on page 385.

RFC 2833 DTMF MTP Passthrough

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 the "Enabling DTMF Integration Using RFC 2833" section on page 390.

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 the"SCCP: Configuring a Voice Mailbox Pilot Number" section on page 382 or "SIP: Configuring a Directory Number for MWI" section on page 397.

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 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 informations, see the "SCCP: Configuring a Phone for MWI Outcall" section on page 395.

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 the "Enabling SIP MWI Prefix Specification" section on page 401.

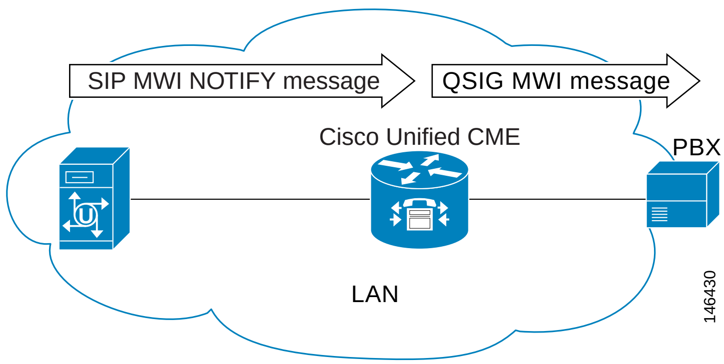
SIP MWI - QSIG Translation

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 17, 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.

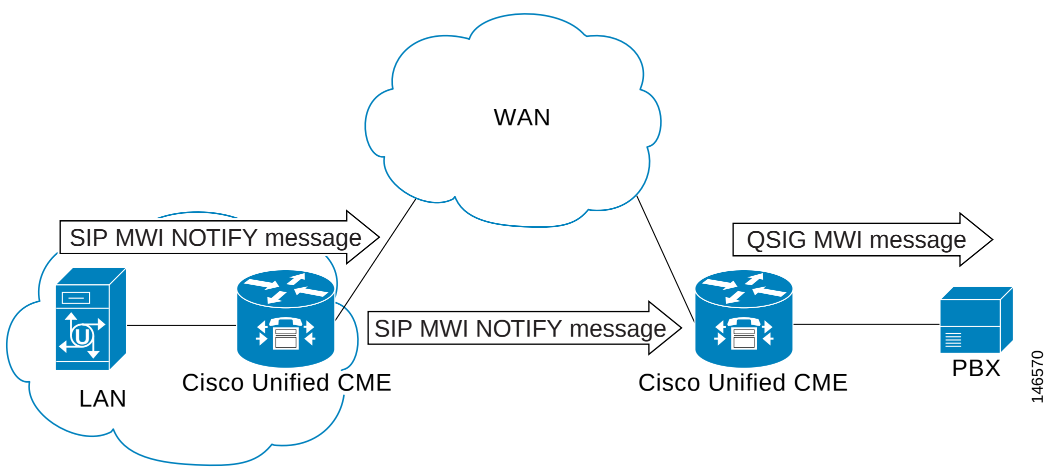
Figure 17 SIP MWI to ISDN QSIG When Voice Mail and Cisco Router are On the Same LAN



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 18, 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 18 SIP MWI to ISDN QSIG When PBX is Connected to a Remote Cisco Router



How to Configure Voice-Mail Integration

This section contains the following tasks:

- SCCP: Configuring a Voice Mailbox Pilot Number, page 382 (required)
- SCCP: Configuring a Mailbox Selection Policy, page 383 (optional)
- SIP: Configuring a Voice Mailbox Pilot Number, page 386 (required)
- Enabling DTMF Integration, page 388 (required)
- SCCP: Configuring a Phone for MWI Outcall, page 395 (optional)
- SIP: Enabling MWI at the System-Level, page 396 (required)
- SIP: Configuring a Directory Number for MWI, page 397 (required)

- Enabling SIP MWI Prefix Specification, page 401 (optional)
- Verifying Voice-Mail Integration, page 401 (optional)

SCCP: Configuring a Voice Mailbox Pilot Number

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.

Prerequisites

- Voicemail phone number must be a valid number; directory number and number for voicemail phone number must be configured. For configuration information, see "Configuring Phones to Make Basic Calls" on page 165.

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 the "SCCP: Configuring a Mailbox Selection Policy" section on page 383.
- To set up DTMF integration patterns for connecting to analog voice-mail applications, see the "Enabling DTMF Integration for Analog Voice-Mail Applications" section on page 388.
- 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 the "Enabling DTMF Integration Using RFC 2833" section on page 390.
- To connect to a Cisco Unity Express system, configure a nonstandard SIP NOTIFY format. See the "Enabling DTMF Integration Using SIP NOTIFY" section on page 393.

SCCP: Configuring a Mailbox Selection Policy

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

- SCCP: Setting a Mailbox Selection Policy for Cisco Unity Express or a PBX Voice-Mail Number, page 383
- SCCP: Setting Mailbox Selection Policy for Cisco Unity, page 385

SCCP: Setting 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.

Prerequisites

Cisco Unified CME 4.0 or a later version.

Restrictions

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.

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 the "Enabling DTMF Integration Using SIP NOTIFY" section on page 393.

SCCP: Setting 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.

Prerequisites

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

Restrictions

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.

SUMMARY STEPS

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

	Command or Action	Purpose
Step 4	**ephone-dn** **Example:** `Router(config)# ephone-dn 752`	Enters ephone-dn configuration mode.
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 the "Enabling DTMF Integration Using RFC 2833" section on page 390.

SIP: Configuring a Voice Mailbox Pilot Number

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.

Prerequisites

- Directory number and number for voicemail phone number must be configured. For configuration information, see "Configuring Phones to Make Basic Calls" on page 165.

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*
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:** `Router(config-register-dn)# call-forward b2bua mailbox 2200`	Designates the voice mailbox to use at the end of a chain of call forwards. • Incoming calls have been forwarded to a busy or no-answer extension will be forwarded to the directory-number specified.

	Command or Action	Purpose
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 the "Enabling DTMF Integration for Analog Voice-Mail Applications" section on page 388.
- 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 the "Enabling DTMF Integration Using RFC 2833" section on page 390.
- To connect to a Cisco Unity Express system, configure a nonstandard SIP NOTIFY format, see the "Enabling DTMF Integration Using SIP NOTIFY" section on page 393.

Enabling DTMF Integration

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

- Enabling DTMF Integration for Analog Voice-Mail Applications, page 388—To set up DTMF integration patterns for connecting to analog voice-mail applications.
- Enabling DTMF Integration Using RFC 2833, page 390—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.
- Enabling DTMF Integration Using SIP NOTIFY, page 393—To configure a SIP dial peer to point to Cisco Unity Express.

Enabling 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 the "SCCP: Configuring a Phone for MWI Outcall" section on page 395.

Enabling 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.

Prerequisites

- Configure the **codec** or **voice-class codec** command for transcoding between G.711 and G.729. See "Configuring Phones to Make Basic Calls" on page 165.

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. **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:** `Router (config-voice-dial-peer)# description CU pilot`	(Optional) Associates a description with the dial peer being configured. Enter a string of up to 64 characters.

	Command or Action	Purpose
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	**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 the "SCCP: Configuring a Phone for MWI Outcall" section on page 395.

Enabling DTMF Integration Using SIP NOTIFY

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

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.
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.

	Command or Action	Purpose
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.
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 the "SCCP: Configuring a Phone for MWI Outcall" section on page 395.

SCCP: Configuring a 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.

Prerequisites

- Directory number and number for MWI line must be configured. For configuration information, see "Configuring Phones to Make Basic Calls" on page 165.

Restrictions

- 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.

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:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	`ephone` *`phone-tag`* **Example:** `Router(config)# ephone 36`	Enters ephone configuration mode.

	Command or Action	Purpose
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 "SCCP: Enabling Ephone-dn Templates" on page 930.
Step 9	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

SIP: Enabling MWI at the System-Level

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

Prerequisites

- 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.

SIP: Configuring a Directory Number for MWI

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.

- SIP: Defining Pilot Call Back Number for MWI Outcall, page 398
- SIP: Configuring a Directory Number for MWI NOTIFY, page 399

SIP: Defining 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.

Prerequisites

- Cisco CME 3.4 or a later version.
- Directory number and number for receiving MWI must be configured. For configuration information, see "Configuring Phones to Make Basic Calls" on page 165.

Restrictions

- 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.

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.

SIP: Configuring 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.

Prerequisites

- 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 "Enabling H.450.7 and QSIG Supplementary Services at a System-Level" on page 553 or "Enabling H.450.7 and QSIG Supplementary Services on a Dial Peer" section on page 554.
- Directory number and number for receiving MWI must be configured. For configuration information, see "Configuring Phones to Make Basic Calls" on page 165.

Restrictions

- 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.

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.

Enabling 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.

Prerequisites

- Cisco Unified CME 4.0 or a later version.
- Directory number for receiving MWI Unsolicited NOTIFY must be configured. For information, see "SIP: Configuring a Directory Number for MWI NOTIFY" section on page 399.

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.

Verifying 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

This section contains the following examples:

Enabling DTMF Integration for Legacy Voice-Mail Applications: Example

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 *
```

Enabling Mailbox Selection Policy for SCCP Phones: Example

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
```

Configuring a Voice Mailbox: Example

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
```

Enabling DTMF Integration Using RFC 2833: Example

The following example shows the configuration for a DTMF Relay:

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

Enabling DTMF Integration Using SIP Notify: Example

The following example shows the configuration for a DTMF Relay:

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

Configuring a SCCP Phone Line for MWI: Example

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
```

Enabling SIP MWI Prefix Specification: Example

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
```

Configuring SIP Directory Number for MWI Outcall: Example

The following example shows an MWI callback pilot number:

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

Configuring a SIP Directory Number for MWI Unsolicited Notify: Example

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
```

Configuring a SIP Directory Number for MWI Subscribe/NOTIFY: Example

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
```

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	• Quick Reference Cards • User Guides

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. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password. If you have a valid service contract but do not have a user ID or password, you can register on Cisco.com.	http://www.cisco.com/techsupport

Feature Information for Voice-Mail Integration

Table 22 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/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

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 22 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 22 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.
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.
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.