Integrating Voice Mail with Cisco Unified SRST

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This chapter describes how to make your existing voice-mail system run on phones connected to a Cisco Unified (SRST) router during Cisco Unified Communications Manager fallback.

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Information About Integrating Voice Mail with Cisco Unified SRST

Cisco Unified SRST can send and receive voice-mail messages from Cisco Unity and other voice-mail systems during Cisco Unified Communications Manager fallback. When the WAN is down, a voice-mail system with BRI or PRI access to the Cisco Unified SRST system uses ISDN signaling (see Figure 1). Systems with Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) access connect to a PSTN and use in-band dual tone multifrequency (DTMF) signaling (see Figure 2).

Figure 1 Cisco Unified Communications Manager Fallback with BRI or PRI

Figure 2 Cisco Unified Communications Manager Fallback with PSTN

Both configurations allow phone message buttons to remain active and calls to busy or unanswered numbers to be forwarded to the dialed numbers' mailboxes.

Calls that reach a busy signal, calls that are unanswered, and calls made by pressing the message button are forwarded to the voice-mail system. To make this happen, you must configure access from the dial peers to the voice-mail system and establish routing to the voice-mail system for busy and unanswered calls and for message buttons.

If the voice-mail system is accessed over FXO or FXS, you must configure instructions (DTMF patterns) for the voice-mail system so that it can access the correct voice-mail system mailbox. If your voice-mail system is accessed over BRI or PRI, no instructions are necessary because the voice-mail system can log in to the calling phone's mailbox directly.

How to Integrate Voice Mail with Cisco Unified SRST

This section contains the following tasks:

- Configuring Direct Access to Voice Mail, page 193 (Required)
- Configuring Message Buttons, page 196 (Required)
- Redirecting to Cisco Unified Communications Manager Gateway, page 198 (Required for BRI or PRI))
- Configuring Call Forwarding to Voice Mail, page 198 (Required FXO or FXS)
- Configuring Message Waiting Indication, page 202 (Optional)

Configuring Direct Access to Voice Mail

To access voice-mail messages with FXO or FXS access, you must have POTS dial peers configured with a destination pattern that matches the voice-mail system's number. Also, you must associate the dial peer with the port to which the voice-mail system is accessed.

Both sets of configurations are done in global configuration mode and in dial-peer configuration mode. The summary and detailed steps below include only the basic commands necessary to perform this task. You may require additional commands for your particular dial-peer configuration.

SUMMARY STEPS

1. **dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**}
2. **destination-pattern** [**+**] *string* [**T**]
3. **port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-no*}
4. **forward-digits** {*num-digit* | **all** | **extra**}
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**} **Example:** `Router(config)#` **`dial-peer voice 1002 pots`**	(FXO or FXS and BRI or PRI) Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. The **dial-peer** command provides different syntax for individual routers. This example is syntax for Cisco 3600 series routers. • *tag*: Digits that define a particular dial peer. Range is from 1 to 2147483647. • **pots**: Indicates that this is a POTS dial peer that uses VoIP encapsulation on the IP backbone. • **voatm**: Specifies that this is a VoATM dial peer that uses real-time AAL5 voice encapsulation on the ATM backbone network. • **vofr**: Specifies that this is a VoFR dial peer that uses FRF.11 encapsulation on the Frame Relay backbone network. • **voip**: Indicates that this is a VoIP dial peer that uses voice encapsulation on the POTS network.
Step 2	**destination-pattern** [**+**] *string* [**T**] **Example:** `Router(config-dial-peer)# destination-pattern 1100T`	(FXO or FXS and BRI or PRI) Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. • **+**: (Optional) Character that indicates an E.164 standard number. • *string*: See Table 1. • **T**: (Optional) Control character that indicates that the destination-pattern value is a variable-length dial string.
Step 3	**port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-no*} **Example:** `Router(config-dial-peer)# port 1/1/1`	(FXO or FXS and BRI or PRI) Associates a dial peer with a specific voice port on Cisco 3600 series routers. • *slot-number*: Number of the slot in the router in which the voice interface card (VIC) is installed. Valid entries are from 0 to 3, depending on the slot in which it is installed. • *subunit-number*: Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1. • *port*: Voice port number. Valid entries are 0 and 1. • *ds0-group-no*: Specifies the DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

	Command or Action	Purpose
Step 4	**forward-digits** {*num-digit* | **all** | **extra**} **Example:** `Router(config-dial-peer)# forward-digits all`	(Optional for FXO or FXS) Specifies which digits to forward for voice calls. • *num-digit*: The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. Range is 0 to 32. Setting the value to 0 is equivalent to entering the **no forward-digits** command. • **all**: Forwards all digits. If **all** is entered, the full length of the destination pattern is used. • **extra**: If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length and ends with the character “T” (for example: T, 123T, 123...T), extra digits are not forwarded.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	(FXO or FXS and BRI or PRI) Exits dial-peer configuration mode.

Table 1 Valid Entries for the String Argument in the destination-pattern command

Entry	Description
Digits 0 to 9	—
Letters A through D	—
Asterisk (*) and pound sign (#)	These appear on standard touch-tone dial pads.
Comma (,)	Inserts a pause between digits.
Period (.)	Matches any entered digit (this character is used as a wildcard).
Percent sign (%)	Indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.
Plus sign (+)	Indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number.
Circumflex (^)	Indicates a match to the beginning of the string. Parentheses (( )), which indicate a pattern and are the same as the regular expression rule.
Dollar sign ($)	Matches the null string at the end of the input string.
Backslash symbol (\)	Is followed by a single character and matches that character. Can be used with a single character with no other significance (matching that character).
Question mark (?)	Indicates that the preceding digit occurred zero or one time.
Brackets ([ ])	Indicates a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.

Examples

The following FXO and FXS example sets up a POTS dial peer named 1102, matches dial-peer 1102 to voice-mail extension 1101, and assigns dial-peer 1102 to voice-port 1/1/1 where the voice-mail system is connected. Other dial peers are configured for direct access to voice mail.

```
voice-port 1/1/1
 timing digit 250
 timing inter-digit 250

dial-peer voice 1102 pots
 destination-pattern 1101
 port 1/1/1
 forward-digits all

dial-peer voice 1103 pots
 destination-pattern 1101
 port 1/1/1
 forward-digits all

dial-peer voice 1104 pots
 destination-pattern 1101
 port 1/1/1
 forward-digits all
```

The following example sets up a POTS dial peer named 1102 to go directly to 1101 through port 2/0:23.

```
controller T1 2/0
 framing esf
 clock source line primary
 linecode b8zs
 cablelength short 133
 pri-group timeslots 21-24

interface Serial2/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn T309-enable
 no cdp enable

voice-port 2/0:23

dial-peer voice 1102 pots
 destination-pattern 1101T
 port 2/0:23
```

Configuring Message Buttons

To activate the message buttons on Cisco Unified IP phones connected to the Cisco Unified SRST router during Cisco Unified Unified Communications Manager fallback, you must program a speed-dial number to the voice-mail system. The speed-dial number is dialed when message buttons on phones connected to the Cisco Unified SRST router are pressed during Cisco Unified Communications Manager fallback. In addition, call forwarding must be configured so that calls to busy and unanswered numbers are sent to the voice-mail number.

This configuration is required for FXO or FXS and BRI or PRI.

SUMMARY STEPS

1. **call-manager-fallback**
2. **voicemail** *phone-number*
3. **call-forward busy** *directory-number*
4. **call-forward noan** *directory-number* **timeout** *seconds*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`call-manager-fallback`** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**`voicemail`** *`phone-number`* **Example:** `Router(config-cm-fallback)#` **`voicemail 5550100`**	Configures the telephone number that is dialed when the message button on a Cisco Unified IP Phone is pressed. • *phone-number*: Phone number configured as a speed-dial number for retrieving messages.
Step 3	**`call-forward busy`** *`directory-number`* **Example:** `Router(config-cm-fallback)# call-forward busy 2000`	Configures call forwarding to another number when the Cisco IP phone is busy. • *directory-number*: Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.
Step 4	**`call-forward noan`** *`directory-number`* **`timeout`** *`seconds`* **Example:** `Router(config-cm-fallback)# call-forward noan 2000 timeout 10`	Configures call forwarding to another number when no answer is received from the Cisco IP phone. • *directory-number*: Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension. • **timeout** *seconds*: Sets the waiting time, in seconds, before the call is forwarded to another phone. The *seconds* range is from 3 to 60000.
Step 5	**`exit`** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Examples

The following example specifies 1101 as the speed-dial number that is issued when message buttons are pressed on Cisco Unified IP Phones connected to the Cisco Unified SRST router. All busy and unanswered calls are configured to be forwarded to the voice-mail number (1101).

```
call-manager-fallback
 voicemail 1101
 call-forward busy 1101
 call-forward noan 1101 timeout 3
```

Redirecting to Cisco Unified Communications Manager Gateway

Note The following task is required for voice-mail systems with BRI or PRI access.

In addition to supporting message buttons for retrieving personal messages, Cisco Unified SRST allows the automatic forwarding of calls to busy and unanswered numbers to voice-mail systems. Voice-mail systems with BRI or PRI access can log in to the calling phone's mailbox directly. For this to happen, some Cisco Unified Communications Manager configuration is recommended. If your voice-mail system supports Redirected Dialed Number Identification Service (RDNIS), RDNIS must be included in the outgoing SETUP message to Cisco Unified Communications Manager to declare the last redirected number and the originally dialed number to and from configured devices and applications.

Step 1 From any page in Cisco Unified Communications Manager, click **Device** and **Gateway.**

Step 2 From the Find and List Gateways page, click **Find**.

Step 3 From the Find and List Gateways page, choose a device name.

Step 4 From the Gateway Configuration page, check **Redirecting Number IE Delivery - Outgoing**.

Configuring Call Forwarding to Voice Mail

Note The following task is required for voice-mail systems with FXO or FXS access.

In addition to supporting message buttons for retrieving personal messages, Cisco Unified SRST allows the automatic forwarding of calls to busy or unanswered numbers to voice-mail systems. The forwarded calls can be routed to almost any location in the voice-mail system. Typically, calls are forwarded to a location in the called number's mailbox where the caller can leave messages.

Call Routing Instructions Using DTMF Digit Patterns

Cisco Unified SRST call-routing instructions are required so that forwarded calls can be sent to the correct voice mailboxes. These instructions consist of DTMF digits configured in patterns that match the dial sequences required by the voice-mail system to get to a particular voice-mail location. For example, a voice-mail system may be designed so that callers must do the following to leave a message:

1. Dial the central voice-mail number (1101) and press #.
2. Dial an extension number (6000) and press #.
3. Dial 2 to select the menu option for leaving messages in the extension number's mailbox.

For Cisco Unified SRST to forward a call to a busy or unanswered number to extension 6000's mailbox, it must be programmed to issue a sequence of 1101#6000#2. As shown in Figure 3, this is accomplished through the **voicemail** and **pattern** commands.

Figure 3 How Voice-Mail Dial Sequence 1101#6000#2 Is Configured in Cisco Unified SRST

```
call-manager-fallback
  voicemail  1101
```

1101 **#6000#2**

```
call-manager-fallback
 pattern ext-to-ext busy # cgn #2
 pattern ext-to-ext busy # cdn #2
 pattern ext-to-ext busy # fdn #2
 pattern ext-to-ext no-answer # cgn #2
 pattern ext-to-ext no-answer # cdn #2
 pattern ext-to-ext no-answer # fdn #2
 pattern trunk-to-ext busy # cgn #2
 pattern trunk-to-ext busy # cdn #2
 pattern trunk-to-ext busy # fdn #2
 pattern trunk-to-ext no-answer # cgn #2
 pattern trunk-to-ext no-answer # cdn #2
 pattern trunk-to-ext no-answer # fdn #2
```

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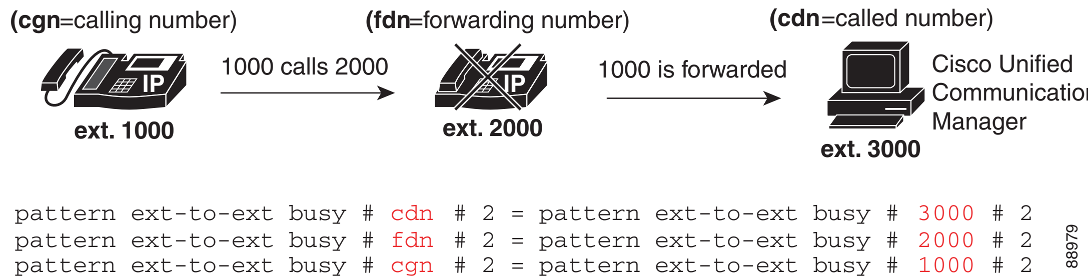
The # cgn #2, # cdn #2, and # fdn #2 portions of the **pattern** commands shown in Figure 3 are DTMF digit patterns. These patterns are composed of tags and tokens. Tags are sets of characters representing DTMF tones. Tokens consist of three command keywords (**cgn**, **cdn**, and **fdn**) that declare the state of an incoming call transferred to voice mail.

A tag can be up to three character from the DTMF tone set (A to D, 0 to 9, # and *). Voice-mail systems can use limited sets of DTMF tones. For example, Cisco Unity uses all DTMF tones but A to D. Tones can be defined in multiple ways. For example, when the star (*) is placed in front of a token by itself, it can mean "dial the following token number," or, if it is at the end of a token, it can mark the end of a token number. If the asterisk is between other tag characters, it can mean dial *. The use of tags depends on how DTMF tones are defined by your voice-mail system.

Tokens tell Cisco Unified SRST what telephone number in the call forwarding chain to use in the pattern. As shown in Figure 4, there are three types of tokens that correspond to three possible call states during voice-mail forwarding.

Figure 4 How Numbers Are Extracted from Tokens



Sets of tags and tokens or patterns activate a voice-mail system when

- A user presses the message button on a phone (**pattern direct** command).
- An internal extension attempts to connect to a busy extension and the call is forwarded to voice mail (**pattern ext-to-ext busy** command).
- An internal extension fails to connect to an extension and the call is forwarded to voice mail (**pattern ext-to-ext no-answer** command).
- An external trunk call reaches a busy extension and the call is forwarded to voice mail (**pattern trunk-to-ext busy** command).
- An external trunk call reaches an unanswered extension and the call is forwarded to voice mail (**pattern trunk-to-ext no-answer** command).

Prerequisites

- FXO hairpin-forwarded calls to voice-mail systems must have disconnect supervision from the central office. For further information, see the *FXO Answer and Disconnect Supervision* document.
- To configure patterns that your voice-mail system will interpret correctly, you must know how the system routes voice-mail calls and interprets DTMF tones (see the "Call Routing Instructions Using DTMF Digit Patterns" section on page 199).

  You can find information about how Cisco Unity handles voice-mail calls in the *How to Transfer a Caller Directly into a Cisco Unity Mailbox* document. Additional call-handling information can be found in the "Subscriber and Operator Orientation" chapters of any Cisco Unity system administration guide.

  For other voice-mail systems, see the analog voice mail integration configuration guide or information about the system's call handling.

SUMMARY STEPS

1. **vm-integration**
2. **pattern direct** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
3. **pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
4. **pattern ext-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
5. **pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

6. **pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`vm-integration`** **Example:** `Router(config)# vm-integration`	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail systems.
Step 2	**`pattern direct`** *`tag1`* {**`CGN`** | **`CDN`** | **`FDN`**} [*`tag2`* {**`CGN`** | **`CDN`** | **`FDN`**}] [*`tag3`* {**`CGN`** | **`CDN`** | **`FDN`**}] [*`last-tag`*] **Example:** `Router(config-vm-int)# 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. • *tag1*: 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. • *tag2* and *tag3*: (Optional) See *tag1*. • *last-tag*: See *tag1*. This tag indicates the end of the pattern. • **CGN**: Calling number (CGN) information is sent to the voice-mail system. • **CDN**: Called number (CDN) information is sent to the voice-mail system. • **FDN**: Forwarding number (FDN) information is sent to the voice-mail system.
Step 3	**`pattern ext-to-ext busy`** *`tag1`* {**`CGN`** | **`CDN`** | **`FDN`**} [*`tag2`* {**`CGN`** | **`CDN`** | **`FDN`**}] [*`tag3`* {**`CGN`** | **`CDN`** | **`FDN`**}] [*`last-tag`*] **Example:** `Router(config-vm-int)# pattern ext-to-ext busy 7 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.
Step 4	**`pattern ext-to-ext no-answer`** *`tag1`* {**`CGN`** | **`CDN`** | **`FDN`**} [*`tag2`* {**`CGN`** | **`CDN`** | **`FDN`**}] [*`tag3`* {**`CGN`** | **`CDN`** | **`FDN`**}] [*`last-tag`*] **Example:** `Router(config-vm-int)# pattern ext-to-ext no-answer 5 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.

	Command or Action	Purpose
Step 5	**pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern trunk-to-ext busy 6 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.
Step 6	**pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# 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. For argument and keyword information, see Step 2.

Examples

For the following configuration, if the voice-mail number is 1101, and 3001 is a phone with a message button, 1101*3001 would be dialed automatically when the 3001 message button is pressed. Under these circumstances, 3001 is considered to be a calling number or inbound call number.

```
vm-integration
 pattern direct * CGN
```

For the following configuration, if 3001 calls 3006 and 3006 does not answer, the SRST router will forward 3001 to the voice-mail system (1101) and send to the voice-mail system the DTMF pattern # 3006 #2. This pattern is intended to select voice mailbox number 3006 (3006’s voice mailbox). For this pattern to be sent, 3001 must be a forwarding number.

```
vm-integration
 pattern ext-to-ext no-answer # FDN #2
```

For the following configuration, if 3006 is busy and 3001 calls 3006, the SRST router will forward 3001 to the voice-mail system (1101) and send to the voice-mail system the DTMF pattern # 3006 #2. This pattern is intended to select voice mailbox number 3006 (3006’s voice mailbox). For this pattern to be sent, 3001 must be a forwarding number.

```
vm-integration
 pattern ext-to-ext busy # FDN #2
```

Configuring Message Waiting Indication

The MWI relay mechanism is initiated after someone leaves a voice-mail message on the remote voice-mail message system. MWI relay is required when one Cisco Unity Voice Mail system is shared by multiple Cisco Unified SRST routers. SRST routers use the SIP Subscribe and Notify methods for MWI. See the *Configuring Cisco IOS SIP Configuration Guide* for more information on SIP MWI and the Subscribe and Notify methods. The SRST router that is the SIP MWI relay server acts as the SIP notifier. The other remote routers act as the SIP subscribers.

SUMMARY STEPS

1. **call-manager-fallback**

2. **mwi relay**
3. **mwi reg-e164**
4. **exit**
5. **sip-ua**
6. **mwi-server** {**ipv4:***destination-address* | **dns:***host-name*} [**expires** *seconds*] [port *port*] [**transport** {**tcp** | **udp**}] [**unsolicited**]
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**call-manager-fallback** **Example:** Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	**mwi relay** **Example:** Router(config-cm-fallback)# mwi relay	Enables the SRST router to relay MWI information to remote Cisco IP phones.
Step 3	**mwi reg-e164** **Example:** Router(config-cm-fallback)# mwi reg-e164	Registers E.164 numbers rather than extension numbers with a SIP proxy or registrar.
Step 4	**exit** **Example:** Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.
Step 5	**sip-ua** **Example:** Router(config)# sip-ua	Enters SIP user-agent configuration mode.

	Command	Purpose
Step 6	**mwi-server** {**ipv4:***destination-address* | **dns:***host-name*} [**expires** *seconds*] [**port** *port*] [**transport** {**tcp** | **udp**}] [**unsolicited**] **Example:** `Router(config-sip-ua)# mwi-server ipv4:10.0.2.254`	Configures voice-mail server settings on a voice gateway or user agent. The IP address and port for the SIP-based MWI server should be in the same LAN as the voice-mail server. The MWI server is a Cisco Unified SRST router. Keywords and arguments are as follows: • **ipv4:***destination-address*: IP address of the voice-mail server. • **dns:***host-name*: Host device housing the domain name server that resolves the name of the voice-mail server. The argument should contain the complete hostname to be associated with the target address; for example, **dns:test.cisco.com**. • **expires** *seconds*: Subscription expiration time, in seconds. Range is from 1 to 999999. Default is 3600. • **port** *port*: Port number on the voice-mail server. Default is 5060. • **transport**: Transport protocol to the voice-mail server. Valid values are tcp and udp. Default is UDP. • **unsolicited**: Requires the voice-mail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.
Step 7	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Configuration Examples

This section provides the following configuration examples:

- Configuring Local Voice-Mail System (FXO and FXS): Example, page 205
- Configuring Central Location Voice-Mail System (FXO and FXS): Example, page 205
- Configuring Voice-Mail Access over FXO and FXS: Example, page 206
- Configuring Voice-Mail Access over BRI and PRI: Example, page 207

Configuring Local Voice-Mail System (FXO and FXS): Example

The "Dial-Peer Configuration for Integration of Voice-Mail with Cisco Unified SRST" section of the example below shows a legacy dial-peer configuration for a local voice-mail system. The "Cisco Unified SRST Voice-Mail Integration Pattern Configuration" section must be compatible with your voice-mail system configuration.

```
! Dial-Peer Configuration for Integration of Voice-Mail with Cisco Unified SRST
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
dial-peer voice 102 pots
 preference 1
 destination-pattern 14011
 port 3/0/1
!
dial-peer voice 103 pots
 preference 2
 destination-pattern 14011
 port 3/1/0
!
dial-peer voice 104 pots
 destination-pattern 14011
 port 3/1/1
!
! Cisco Unified SRST configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
 call-forward busy 14011
 call-forward noan 14011 timeout 3

! Cisco Unified SRST Voice-Mail Integration Pattern Configuration
!
vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```

Configuring Central Location Voice-Mail System (FXO and FXS): Example

The "Dial-Peer Configuration for Integration of Voice-Mail with Cisco Unified SRST in Central Location" section of the example shows a legacy dial-peer configuration for a central voice-mail system. The "Cisco Unified SRST Voice-Mail Integration Pattern Configuration" section must be compatible with your voice-mail system configuration.

Note Message waiting indicator (MWI) integration is not supported for PSTN access to voice-mail systems at central locations.

```
! Dial-Peer Configuration for Integration of Voice-Mail with Cisco Unified SRST in Central
```

```
! Location
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
! Cisco Unified SRST configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
 call-forward busy 14011
 call-forward noan 14011 timeout 3
!
! Cisco Unified SRST Voice-Mail Integration Pattern Configuration
!
vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```

Configuring Voice-Mail Access over FXO and FXS: Example

The following example shows how to configure the Cisco Unified SRST router to forward unanswered calls to voice mail. In this example, the voice-mail number is 1101, the voice-mail system is connected to FXS voice port 1/1/1, and the voice mailbox numbers are 3001, 3002, and 3006.

```
voice-port 1/1/1
 timing digit 250
 timing inter-digit 250

dial-peer voice 1102 pots
 destination-pattern 1101T
 port 1/1/1

call-manager-fallback
 timeouts interdigit 5
 ip source-address 1.6.0.199 port 2000
 max-ephones 24
 max-dn 24
 transfer-pattern 3...
 voicemail 1101
 call-forward busy 1101
 call-forward noan 1101 timeout 3
 moh minuet.au

vm-integration
 pattern direct * CGN
 pattern ext-to-ext no-answer # FDN #2
 pattern ext-to-ext busy # FDN #2
 pattern trunk-to-ext no-answer # FDN #2
 pattern trunk-to-ext busy # FDN #2
```

Configuring Voice-Mail Access over BRI and PRI: Example

The following example shows how to configure the Cisco Unified SRST router to forward unanswered calls to voice mail. In this example, the voice-mail number is 1101, the voice-mail system is connected to a BRI or PRI voice port, and the voice mailbox numbers are 3001, 3002, and 3006.

```
controller T1 2/0
 framing esf
 clock source line primary
 linecode b8zs
 cablelength short 133
 pri-group timeslots 21-24

interface Serial2/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn T309-enable
 no cdp enable

voice-port 2/0:23

dial-peer voice 1102 pots
 destination-pattern 1101T
 direct-inward-dial
 port 2/0:23

call-manager-fallback
 timeouts interdigit 5
 ip source-address 1.6.0.199 port 2000
 max-ephones 24
 max-dn 24
 transfer-pattern 3...
 voicemail 1101
 call-forward busy 1101
 call-forward noan 1101 timeout 3
 moh minuet.au
```

Where to Go Next

For information about monitoring and maintaining Cisco Unified SRST, go to the "Monitoring and Maintaining Cisco Unified SRST" section on page 223.

For additional information, see the "Additional References" section on page 44 in the "Overview of Cisco Unified SRST" section on page 31.