Configuring SIP MWI Features

This module describes message-waiting indication (MWI) in a SIP-enabled network.

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Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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.

Prerequisites for SIP MWI

SIP MWI NOTIFY - QSIG MWI Translation Feature

- Ensure that you have a working SIP network with the following:
    - A voice-messaging system that provides a SIP MWI Notify message to the phone--including Cisco Unified Communications Manager (formerly known as Cisco CallManager), Release 5.0 or later or Cisco Unified Communications Manager Express (Cisco Unified CME, formerly known as Cisco CallManager Express) Release 4.0 or later.

    - Voice messaging on Cisco Unity 4.0.1 or later releases (colocated or integrated with the Cisco Unified Communications Manager) or an ISDN Q-signaling (QSIG) PBX.
- Connect gateway and Cisco routers directly to a PBX.
- Ensure that phones connected to PBXs support MWI notification.

SIP Audible Message-Waiting Indicator for FXS Phones Feature

- The MWI tone is generated by the voice-mail server. Be sure that you understand how to configure MWI service on a voice-mail server (such as Cisco Unity).

Restrictions for SIP MWI

SIP MWI NOTIFY - QSIG MWI Translation Feature

- Visual MWI for phones is a functionality of the phone itself and is not addressed in this document.
- The feature supports only SIP unsolicited notify and does not support SIP subscribe notify.
- This feature is not supported in trunk groups in ISDN circuits. In this scenario, trunk groups disable the SIP MWI feature.

SIP Audible Message-Waiting Indicator for FXS Phones Feature

- The SIP Audible Message-Waiting Indicator for FXS Phones feature does not provide the following functionality:
    - Security or authentication services
    - Call redirection to the voice-mail server when the line is busy or there is no answer
    - Instructions on accessing the voice-mail server or retrieving voice messages

Information About SIP MWI

The SIP Audible Message-Waiting Indicator for FXS Phones feature enables an FXS port on a voice gateway to receive audible MWI in a SIP-enabled network. The FXS port on a voice gateway is an RJ-11 connector that allows connections to basic telephone service equipment.

This feature provides the following benefits:

- Message waiting is now indicated to FXS phone users through an audible tone, replicating the functionality users have with traditional telephone systems.
- By means of the Cisco IOS command-line interface, you can enable or disable MWI under the voice port and configure one voice-mail server per user agent (UA) or voice gateway.

To configure SIP MWI support, you should understand the following concepts:

SUBSCRIBE NOTIFY MWI

MWI is a common feature of telephone networks and uses an audible indication (such as a special dial tone) that a message is waiting. The IETF draft A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP) draft-ietf-sipping-mwi-03.txt defines MWI as “a SIP event package carrying message waiting status and message summaries from a messaging system to an interested user agent.”

In Cisco SIP networks, the event notification mechanisms used to carry message waiting status are the SUBSCRIBE and NOTIFY methods. The SUBSCRIBE method requests notification of an event. The NOTIFY method provides notification that an event requested by an earlier SUBSCRIBE method has occurred.

Note For information on the SUBSCRIBE and NOTIFY methods, see the “Configuring Additional SIP Application Support” chapter of the *Cisco IOS SIP Configuration Guide* .

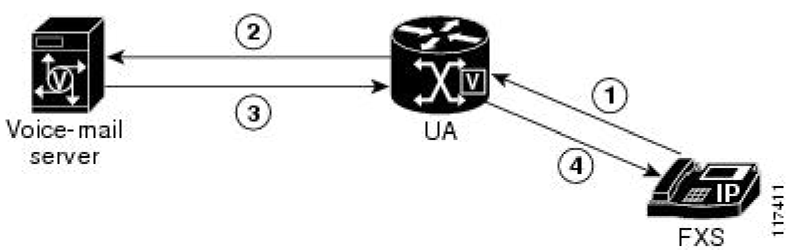
In this feature, a UA (on behalf of the analog FXS phone) subscribes to a voice-mail server to request notification of mailbox status. When the mailbox status changes, the voice-mail server notifies the UA. The UA then indicates that there is a change in mailbox status by providing an MWI tone when the user takes the phone off-hook.

The frequency and cadence of the MWI tone may vary from country to country. For North America, it is defined in GR-506. After you configure the **cp tone** command under your voice port, Cisco IOS software chooses the correct MWI tone accordingly.

Each voice port has its own subscription and notification process. If there are multiple dial peers associated with an FXS voice port, multiple subscriptions are sent to the voice-mail server. If the voice port does not have MWI enabled, the voice gateway returns a 481 Call Leg/Transaction Does Not Exist message to the voice-mail server.

The figure below shows the basic MWI subscription and notification flow.

Figure 1: MWI Notification Flow



1. The user enables the MWI service for the FXS phone by configuring the voice gateway.
2. The UA sends a subscription request to the server on the user’s behalf.
3. The voice-mail server notifies the UA when there is a change in voice-mail status.
4. The UA notifies the phone user with an audible tone.

Unsolicited MWI

In addition to the MWI status forwarded by using the SUBSCRIBE and NOTIFY methods, unsolicited MWI notify is also supported. With unsolicited MWI, MWI service is initially configured on the voice-mail server.

The UA does not need to subscribe to the voice-mail server to receive MWI service. If configured for unsolicited MWI, the voice-mail server automatically sends a SIP notification message to the UA if the mailbox status changes.

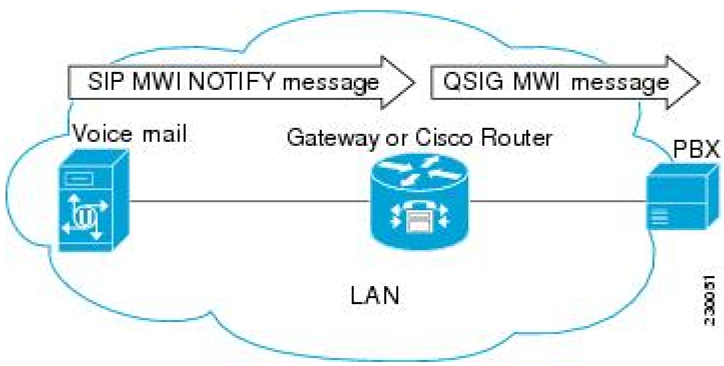
SIP MWI NOTIFY - QSIG MWI Translation

In Cisco IOS Release 12.4(11)T, the SIP MWI NOTIFY - QSIG Translation feature enhances MWI functionality to include SIP-MWI-NOTIFY-to-QSIG-MWI translation between Cisco gateways or routers over a LAN or WAN and extends message waiting indicator (MWI) functionality for SIP MWI and QSIG MWI interoperation to enable sending MWI over QSIG from a Cisco IOS SIP gateway 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 the MWI lamp either on or off on the corresponding IP phone as appropriate.

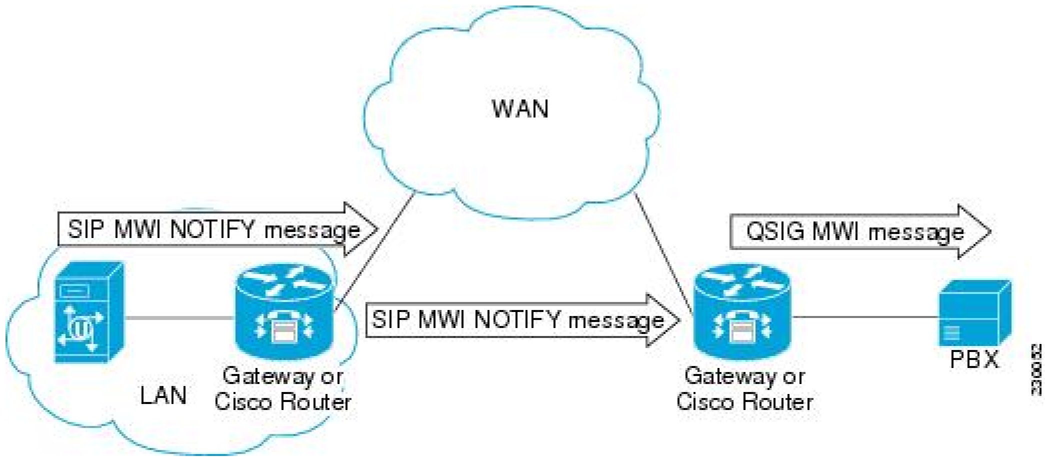
This feature supports only Unsolicited NOTIFY. Subscribe NOTIFY is not supported by this feature.

In the figure below, 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.



Whether the SIP Unsolicited NOTIFY is received via LAN or WAN does not matter as long as the PBX is connected to the gateway or Cisco router, and not to the remote voice mail server.

In the figure below, a voice mail system, such as Cisco Unity, and Unified CME are connected to the same LAN and a remote Unified CME is connected across the WAN. In this scenario, the protocol translation is performed at the remote Unified CME router and the QSIG MWI message is sent to the PBX.



How to Configure SIP MWI

This section contains the following procedures for configuring the SIP Audible MWI for FXS Phones feature:

Note For help with a procedure, see the verification and troubleshooting sections listed above. Before you perform a procedure, familiarize yourself with the following information:

Configuring SIP MWI NOTIFY - QSIG MWI Translation

This section contains information for configuring SIP MWI NOTIFY - QSIG MWI Translation on a gateway.

Note All configuration for this feature is done on the gateway or Cisco router.

Configuring the Gateway

To configure SIP MWI NOTIFY - QSIG MWI Translation on a gateway, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot* / *port*
4. **mwi**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot* / *port* **Example:** `Router(config)# voice-port 2/2`	Enters voice-port configuration mode for the specified PRI or BRI voice port.
Step 4	**mwi** **Example:** `Router (config-voiceport)# mwi`	Enables MWI on this voice port. **Note** If the voice port is not configured for MWI, the gateway returns a 481 Call Leg/Transaction Does Not Exist message to the voice-mail server. If multiple dial peers are associated with the same voice port, multiple subscriptions are sent to the voice-mail server.
Step 5	**exit** **Example:** `Router(config-dial-peer-voice)# exit`	Exits the current configuration mode.

Configuring Voice-Mail Server Settings on the UA

Note This configuration initiates the capability of a UA or voice gateway to indicate voice-mail status changes. One voice-mail server is configured per voice gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server** {**ipv4:** *destination-address* | **dns :** *host-name*} [**expires***seconds*] [**port***port*] [**transport** {**tcp** | **udp**}] [**unsolicited**]
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode. Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**mwi-server** {**ipv4:** *destination-address* | **dns :** *host-name*} [**expires***seconds*] [**port***port*] [**transport** {**tcp** | **udp**}] [**unsolicited**] **Example:** `Router(config-sip-ua)# mwi-server dns:test.example.com expires 86000 port 5060 transport udp unsolicited`	Configures voice-mail server settings on a voice gateway or UA. 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.example.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 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.

Configuring the Voice-Mail Server for Unsolicited

To configure the Cisco Unity voice-mail server to be unsolicited, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server ipv4:** x.x.x.x **unsolicited**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP-user-agent configuration mode.
Step 4	**mwi-server ipv4:** x.x.x.x **unsolicited** **Example:** `Router (config-sip-ua)# mwi-server ipv4:192.0.10.150 unsolicited`	Configures the specified voice-mail (MWI) server to be unsolicited. (That is, requires the server to send a SIP notification message to the voice gateway or user agent if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.)
Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current configuration mode.

Enabling MWI Under an FXS Voice Port

To enable MWI under the specified FXS voice port, perform the following steps.

Note If the voice port does not have MWI enabled, the voice gateway returns a 481 Call Leg/Transaction Does Not Exist message to the voice-mail server.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port*
4. **cptone** *locale*
5. **mwi**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters 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 2/2`	Enters voice-port configuration mode. To find the *port* argument for your router, see the Cisco IOS Voice Command Reference*,* Release 12.3T.
Step 4	**cptone** *locale* **Example:** `Router(config-voiceport)# cptone us`	Specifies a regional analog voice-interface-related tone, ring, and cadence setting for a specified FXS voice port.
Step 5	**mwi** **Example:** `Router(config-voiceport)# mwi`	Enables MWI for a specified FXS voice port.
Step 6	**exit** **Example:** `Router(config-voiceport)# exit`	Exits the current mode.

Verifying MWI Settings

SUMMARY STEPS

1. **show sip-ua mwi**

DETAILED STEPS

show sip-ua mwi

Use this command to display SIP MWI settings from the voice-mail server. The command displays endpoint status as OFF if a message is deleted or if no message is waiting. The endpoint status changes to ON when a message is waiting.

The following sample output shows endpoint status as OFF if a message is deleted or if no message is waiting. The endpoint status changes to ON when a message is waiting.

Example:

```
Router#
show sip-ua mwi
MWI type: 2
MWI server: dns:unity-vm.example1.com
MWI expires: 60
MWI port: 5060
MWI transport type: UDP
MWI unsolicited
MWI server IP address:
C801011E
0
0
0
0
0
0
0
MWI ipaddr cnt 1:
MWI ipaddr idx 0:
MWI server: 192.168.1.30, port 5060, transport 1
MWI server dns lookup retry cnt: 0
endpoint 8000  mwi status ON
endpoint 8000  mwi status ON
endpoint 8001  mwi status OFF
```

Configuring VMWI on analog phones connected to FXS

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.

To enable MWI under the specified FXS voice port, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port*
4. **mwi**
5. Do one of the following:
   - **vmwi dc-voltage**
   -
   -
   - **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.

	Command or Action	Purpose
Step 5	Do one of the following: • **vmwi dc-voltage** • • • **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.

Step 8	**mwi-server** {**ipv4**:*destination-address* | **dns**:*host-name*} [**unsolicited**]
	Example:
	`Router(config-sip-ua)# mwi-server ipv4:1.5.49.200`
	Example:
	`or`
	Example:
	`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.

What to do next

Troubleshooting Tips

Note For general troubleshooting tips and a list of important **debug** commands, see the "Verifying and Troubleshooting SIP Features" chapter in the *Cisco IOS SIP Configuration Guide*.

- Use the **debug ccsip messages** command for debugging purposes.
- Use the debug vpm all command for showing the VMWI state of a voice-port

Following is sample output for this command:

Sample Output for the debug ccsip messages Command

The following sample output is from the perspective of a SIP UA acting on the behalf of an analog FXS phone. The output shows that when the phone connected to the UA is called and the line is busy, the caller leaves a message. The UA, connected to the voice-mail server, receives notification and provides a tone to the user. The user listens to the message and deletes it.

```
Router# debug ccsip messages
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:78002@csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK24E9
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Supported: 100rel,timer
Min-SE:  1800
Cisco-Guid: 3659524871-1844515286-2148452871-566800187
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Remote-Party-ID: "SIPMWI-1" <sip:78001@192.168.1.174>;party=calling;screen=no;privacy=off
Timestamp: 1022206059
Contact: <sip:78001@192.168.1.174:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 234
v=0
o=CiscoSystemsSIP-GW-UserAgent 5421 615 IN IP4 192.168.1.174
s=SIP Call
c=IN IP4 192.168.1.174
t=0 0
m=audio 16818 RTP/AVP 18 19
c=IN IP4 192.168.1.174
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
a=ptime:20
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
```

```
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK24E9
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
CSeq: 101 INVITE
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK24E9
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=5ea400de-695763f1
CSeq: 101 INVITE
Proxy-Authenticate: DIGEST realm="example.com", nonce="40871b34", qop="auth", algorithm=MD5
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:78002@csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK24E9
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=5ea400de-695763f1
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Max-Forwards: 70
CSeq: 101 ACK
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:78002@csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Supported: 100rel,timer
Min-SE:  1800
Cisco-Guid: 3659524871-1844515286-2148452871-566800187
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq: 102 INVITE
Max-Forwards: 70
Remote-Party-ID: "SIPMWI-1" <sip:78001@192.168.1.174>;party=calling;screen=no;privacy=off
Timestamp: 1022206059
Contact: <sip:78001@192.168.1.174:5060>
Expires: 180
Allow-Events: telephone-event
Proxy-Authorization: Digest
username="user1",realm="example.com",uri="sip:192.168.1.37",response="df92654ce55d734
6398013442919e7fc",nonce="40871b34",cnonce="2AEBD5CD",qop=auth,algorithm=MD5,nc=00000001
Content-Type: application/sdp
Content-Length: 234
v=0
o=CiscoSystemsSIP-GW-UserAgent 5421 615 IN IP4 192.168.1.174
s=SIP Call
c=IN IP4 192.168.1.174
t=0 0
m=audio 16818 RTP/AVP 18 19
c=IN IP4 192.168.1.174
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
```

```
a=rtpmap:19 CN/8000
a=ptime:20
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
CSeq: 102 INVITE
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:78002@192.168.1.174:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1
Record-Route:
<sip:78001.474b6083-19c218c7-16e9de49-93b83d71@192.168.1.174:5060;maddr=192.168.1.37>
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Supported: 100rel,timer
Min-SE: 1800
Cisco-Guid: 3659524871-1844515286-2148452871-566800187
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq: 102 INVITE
Max-Forwards: 69
Remote-Party-ID: "SIPMWI-1" <sip:78001@192.168.1.174>;party=calling;screen=no;privacy=off
Timestamp: 1022206059
Contact: <sip:78001@192.168.1.174:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 234
v=0
o=CiscoSystemsSIP-GW-UserAgent 5421 615 IN IP4 192.168.1.174
s=SIP Call
c=IN IP4 192.168.1.174
t=0 0
m=audio 16818 RTP/AVP 18 19
c=IN IP4 192.168.1.174
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
a=ptime:20
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1,SIP/2.0/UDP
 192.168.1.174:5060;re
ceived=192.168.1.174;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A843C-187B
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Timestamp: 1022206059
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 INVITE
Allow-Events: telephone-event
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
```

```
Sent:
SIP/2.0 486 Busy here
Via: SIP/2.0/UDP 192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1,SIP/2.0/UDP
 192.168.1.174:5060;re
ceived=192.168.1.174;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A843C-187B
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Timestamp: 1022206059
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 INVITE
Allow-Events: telephone-event
Reason: Q.850;cause=17
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:78002@192.168.1.174:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A843C-187B
CSeq: 102 ACK
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 180 Ringing
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
CSeq: 102 INVITE
Content-Length: 0
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
Record-Route:
<sip:7200@example1.com:5060;maddr=192.168.1.37>,<sip:78002@csps-release.example1.com:5060;maddr=192.168.1.37>
Contact: sip:7200@192.168.1.30:5060
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
CSeq: 102 INVITE
Content-Length: 166
Content-Type: application/sdp
v=0
o=192.168.1.30 7542610 7542610 IN IP4 192.168.1.30
s=No Subject
c=IN IP4 192.168.1.30
t=0 0
m=audio 22840 RTP/AVP 18
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:78002@csps-release.example1.com:5060;maddr=192.168.1.37 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK10EF
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Route: <sip:7200@example1.com:5060;maddr=192.168.1.37>,<sip:7200@192.168.1.30:5060>
Max-Forwards: 70
```

```
CSeq: 102 ACK
Proxy-Authorization: Digest
username="user1",realm="example.com",uri="sip:192.168.1.37",response="631ff1eec9e21b0
2fcbdbe932c9f7b5b",nonce="40871b34",cnonce="81C16CF6",qop=auth,algorithm=MD5,nc=00000002
Content-Length: 0
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
REGISTER sip:csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK171F
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:48 GMT
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1022206068
CSeq: 14 REGISTER
Contact: <sip:78002@192.168.1.174:5060>
Expires: 60
Content-Length: 0
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK171F
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 14 REGISTER
Content-Length: 0
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK171F
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 14 REGISTER
WWW-Authenticate: DIGEST realm="example.com", nonce="40871b3d", qop="auth", algorithm=MD5
Content-Length: 0
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
REGISTER sip:csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK21B5
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:48 GMT
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1022206068
CSeq: 15 REGISTER
Contact: <sip:78002@192.168.1.174:5060>
Expires: 60
Authorization: Digest
username="user2",realm="example.com",uri="sip:192.168.1.37",response="134885a71dd9690370196
089e445e955",nonce="40871b3d",cnonce="7446932B",qop=auth,algorithm=MD5,nc=00000001
Content-Length: 0
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK21B5
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
```

```
CSeq: 15 REGISTER
Content-Length: 0
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK21B5
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 15 REGISTER
Contact: <sip:78002@192.168.1.174:5060>;expires=60
Content-Length: 0
00:11:44: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
BYE sip:78002@csps-release.example1.com:5060;maddr=192.168.1.37 SIP/2.0
Via: SIP/2.0/UDP  192.168.1.174:5060;branch=z9hG4bK79A
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Route: <sip:7200@example1.com:5060;maddr=192.168.1.37>,<sip:7200@192.168.1.30:5060>
Timestamp: 1022206074
CSeq: 103 BYE
Reason: Q.850;cause=16
Proxy-Authorization: Digest
username="user1",realm="example.com",uri="sip:192.168.1.37",response="dffc15fe72d26b9
3d78162852ae1a341",nonce="40871b34",cnonce="AF9FD85E",qop=auth,algorithm=MD5,nc=00000003
Content-Length: 0
00:11:44: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK79A
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
CSeq: 103 BYE
Content-Length: 0
00:11:44: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK79A
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
CSeq: 103 BYE
Content-Length: 0
```

Sample relevant output for the debug vpm all command

```
Process vmwi. vmwi state: OFF
The phone is not onhook (1). Delay the vmwi processing.
Process dc-voltage vmwi. State: OFF
*Mar  2 02:33:34.841: [2/0] c2400_dc_volt_mwi: on=0
The phone is not onhook (1). Delay the vmwi processing.
Process vmwi. vmwi state: ON
```

Configuration Examples for SIP MWI

The following example shows that SIP MWI is configured on the gateway.

```
Router# show running-config
Building configuration...
Current configuration : 14146 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
no service dhcp
!
boot-start-marker
boot system flash:c2430-is-mz.mwi_dns
boot-end-marker
!
card type e1 1
logging buffered 9000000 debugging
!
username all
network-clock-participate E1 1/0
network-clock-participate E1 1/1
no aaa new-model
no ip subnet-zero
!
ip domain name example1.com
ip name-server 192.168.1.1
ip dhcp excluded-address 172.16.224.97
!
isdn switch-type primary-qsig
!
trunk group  Incoming
!
voice-card 0
!
voice service voip
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
 h323
 sip
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g726r32
!
voice hpi capture buffer 100000
voice hpi capture destination flash:t1.dat
!
voice translation-rule 1
 rule 1 /.*/ /8005550100/
!
voice translation-profile Out
 translate calling 1
!
controller E1 1/0
 linecode ami
 pri-group timeslots 1-31
!
controller E1 1/1
```

```
 linecode ami
 pri-group timeslots 1-10,16
!
interface FastEthernet0/0
 ip address 192.168.1.172 255.255.255.0
 no ip mroute-cache
 duplex half
 speed auto
!
interface FastEthernet0/1
 ip address 10.2.141.19 255.255.0.0
 no ip mroute-cache
 duplex auto
 speed auto
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 192.168.1.2
!
ip rtcp report interval 30000
!
control-plane
!
! Enable MWI on voice ports 2/0 and 2/1.
!
voice-port 2/0
 mwi
 timeouts ringing 30
 station-id name SIPUser1
 station-id number 8000
 caller-id enable
!
voice-port 2/1
 mwi
 timeouts ringing 30
 station-id name SIPUser2
 station-id number 8001
 caller-id enable
!
dial-peer cor custom
!
! Configure dial peers.
!
dial-peer voice 1 pots
 preference 5
 destination-pattern 8000
 port 2/0
!
dial-peer voice 2 pots
 preference 5
 destination-pattern 8001
 port 2/1
!
dial-peer voice 3 voip
 destination-pattern .T
 voice-class codec 1
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
!
dial-peer voice 7 pots
 trunkgroup Incoming
 destination-pattern 789...
!
```

```
dial-peer voice 8 pots
 trunkgroup Incoming
 destination-pattern 789...
!
dial-peer voice 22 voip
 destination-pattern 7232
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
 codec g711ulaw
!
gateway
 timer receive-rtcp 5
 timer receive-rtp 1200
!
! Configure the voice-mail server settings on the gateway with the mwi-server command.
!
sip-ua
 authentication username user1 password password1 realm example.com
 mwi-server dns:test.example.com expires 60 port 5060 transport udp unsolicited
 registrar dns:csps-release.test.example.com expires 3600
 sip-server dns:csps-release.test.example.com
!
telephony-service
 max-dn 100
 max-conferences 4
!
ephone-dn  1
!
line con 0
 exec-timeout 0 0
 password 7 password2
 transport preferred all
 transport output all
line aux 0
 transport preferred all
 transport output all
line vty 0 4
 password 7 password3
 login
 transport preferred all
 transport input all
 transport output all
!
end
```

Configuration Example for SIP MWI NOTIFY - QSIG MWI Translation

The following example shows a sample configuration of the SIP MWI NOTIFY - QSIG MWI Translation feature on a SIP gateway.

```
dial-peer voice 1000 voip
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.120.70.10
 incoming called-number .T
 dtmf-relay rtp-nte
!
```

```
sip-ua
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
end
```

Configuration Example for SIP VMWI

```
Router# show running-config
Building configuration...
!
sip-ua
 mwi-server ipv4:9.13.40.83 expires 3600 port 7012 transport udp unsolicited
!
voice-port 2/0
 vmwi dc-voltage
 mwi
!
```

Additional References

General SIP References

- “Basic SIP Configuration” chapter--Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter (listed alphabetically)

- RFC 3842， “A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)” at http://www.ietf.org/rfc/rfc3842.txt
- *Cisco IOS Voice Command Reference*
- *Cisco IOS Voice Configuration Library*

Feature Information for SIP MWI

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 Configuring SIP MWI Features

Feature Name	Releases	Feature Information
SIP Audible Message-Waiting Indicator for FXS Phones	12.3(8)T	This feature enables an FXS port on a voice gateway to receive audible MWI in a SIP-enabled network.
SIP MWI NOTIFY - QSIG MWI Translation	12.4(11)T	This feature was introduced. This feature is used to configure SIP MWI NOTIFY - QSIG MWI Translation on a gateway.
VMWI on analog phones connected to FXS	15.1(2)T	This feature introduces support fpor VMWI on analog phones connected to FXS.