



Integrating Cisco CallManager and Cisco SRST to Use Cisco SRST as a Multicast MOH Resource

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This document describes how to configure Cisco CallManager and Cisco Survivable Remote Site Telephony (SRST) to allow Cisco CallManager to use Cisco SRST gateways as multicast music-on-hold (MOH) resources during fallback and normal Cisco CallManager operation. A distributed MOH design with local gateways providing MOH eliminates the need to stream MOH across a WAN and saves bandwidth.

Feature History for Cisco SRST as a Multicast MOH Resource

| Release | Modification |
|------------|---|
| 12.2(15)ZJ | This feature was introduced as part of Cisco SRST V3.0. |
| 12.3(11)T | The MOH-live feature was added. |

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Prerequisites for Using Cisco SRST Gateways as a Multicast MOH Resource, page 2](#)
- [Restrictions for Using Cisco SRST Gateways as a Multicast MOH Resource, page 3](#)
- [Information About Using Cisco SRST Gateways as a Multicast MOH Resource, page 4](#)
- [How to Use Cisco SRST Gateways as a Multicast MOH Resource, page 8](#)
- [Configurations Examples for Cisco SRST Gateways, page 30](#)
- [Additional References, page 31](#)

Prerequisites for Using Cisco SRST Gateways as a Multicast MOH Resource

- Multicast MOH for H.323 and MGCP is supported on Cisco CallManager 3.1.1 and higher.
- Cisco CallManager must be configured as follows:
 - With multicast MOH enabled.
 - With Media Resource Groups (MRGs) and Media Resource Group Lists (MRGLs) controlling which devices receive multicast MOH and which devices receive unicast MOH.
 - With Cisco CallManager regions assigned in such a way that G.711 is used whenever a Cisco SRST multicast MOH resource is invoked.
- The Cisco SRST gateways must run on Cisco SRST 3.0 on Cisco IOS Release 12.2(15)ZJ2 or later.
- Cisco CallManager and Cisco SRST must run on either H.323 or MGCP.
- For branches that do not run Cisco SRST, Cisco CallManager multicast MOH packets must cross the WAN. To accomplish this, you must have multicast routing enabled in your network. For more information about multicast routing, see the “IP Multicast” section of the [Cisco IOS IP Configuration Guide, Release 12.3](#).

- With Cisco IOS releases prior to 12.3(14)T, configure SRST as your MGCP gateway's fallback mode using the **ccm-manager fallback-mgcp** and **call application alternate** commands. With Cisco IOS releases after 12.3(14)T, the **ccm-manager fallback-mgcp** and **service** commands must be configured. Configuring these two commands allows SRST to assume control over the voice port and over call processing on the MGCP gateway. A complete configuration describing setting up SRST as your fallback mode is shown in the [Cisco IOS Survivable Remote Site Telephony Version 3.3 System Administrator Guide](#).

Restrictions for Using Cisco SRST Gateways as a Multicast MOH Resource

- Cisco SRST multicast MOH does not support unicast MOH.
- Only a single Cisco CallManager audio source can be used throughout the network. However, the audio files on each SRST gateway's flash memory can be different.
- Cisco SRST multicast MOH supports G.711 only.
- Multicast MOH support for H.323 is unavailable in all versions of Cisco CallManager 3.3.2. For more information, see CSCdz00697 using [Bug Toolkit](#).
- In the Cisco IOS Release 12.2(15)ZJ image for Cisco 1700 series gateways, Cisco SRST multicast MOH does not include support for H.323 mode.

Information About Using Cisco SRST Gateways as a Multicast MOH Resource

To configure SRST gateways as an MOH resource, you should understand the following concepts:

- [SRST Gateways and Cisco CallManager, page 4](#)
- [Codecs, Port Numbers, and IP Addresses, page 5](#)
- [Multicast MOH Transmission, page 7](#)
- [SRST MOH Live Feed Support, page 7](#)

SRST Gateways and Cisco CallManager

Cisco SRST gateways can be configured to multicast Real-Time Transport Protocol (RTP) packets from flash memory during fallback and normal Cisco CallManager operation. To make this happen, Cisco CallManager must be configured for multicast MOH in such a way that the audio packets do not cross the WAN. Instead, audio packets are broadcast from the flash memory of Cisco SRST gateways to the same multicast MOH IP address and port number configured for Cisco CallManager multicast MOH. IP phones at remote sites are able to pick up RTP packets that are multicast from the local branch gateways instead of from the central Cisco CallManager.

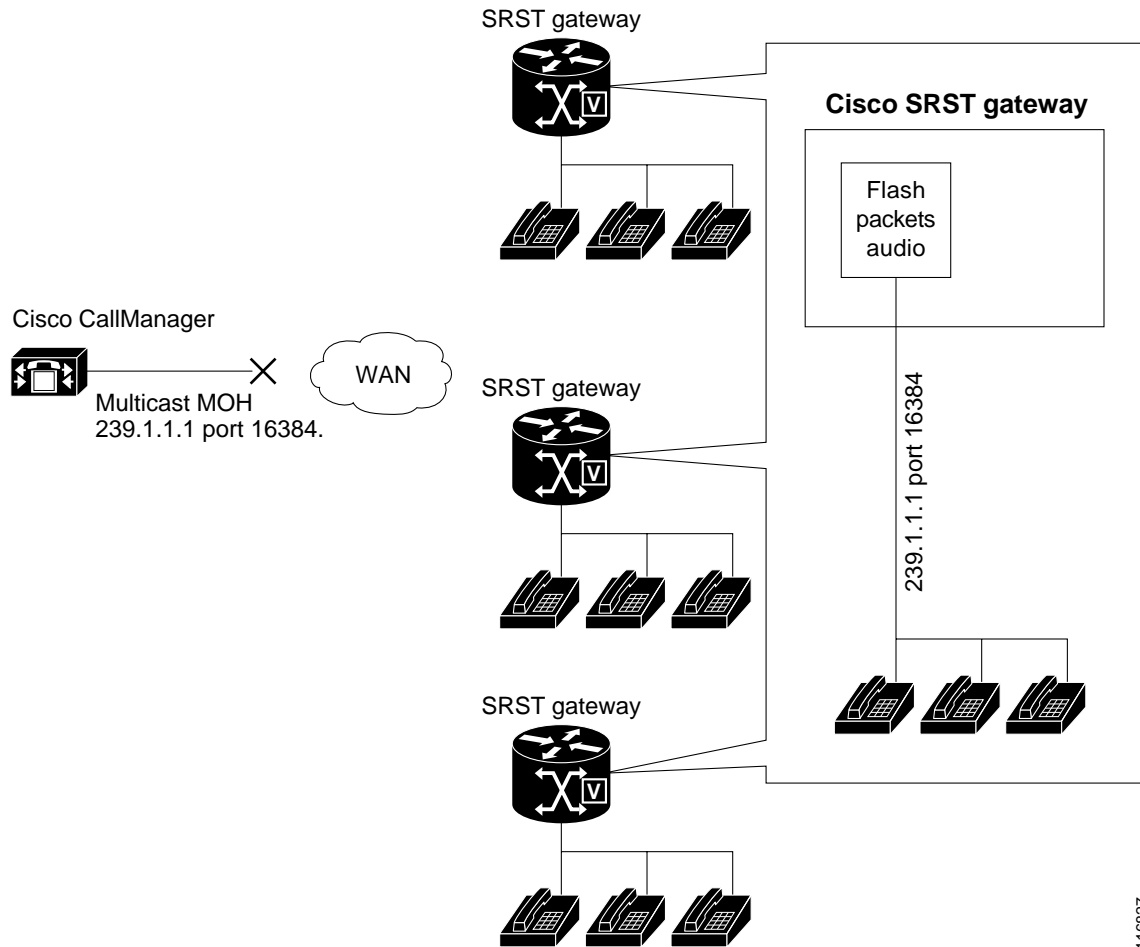
[Figure 1](#) shows a sample configuration in which all phones are configured by Cisco CallManager to receive multicast MOH through port number 16384 and IP address 239.1.1.1. Cisco CallManager is configured in such a way that multicast MOH cannot reach the WAN, and local Cisco SRST gateways are configured to send audio packets from their flash files to port number 16384 and IP address 239.1.1.1. Cisco CallManager and the IP phones are spoofed and behave as if Cisco CallManager were originating the multicast MOH.



Note

Phone users at the central site would use multicast MOH from the central site.

Figure 1 Multicast MOH from Cisco SRST Flash Memory

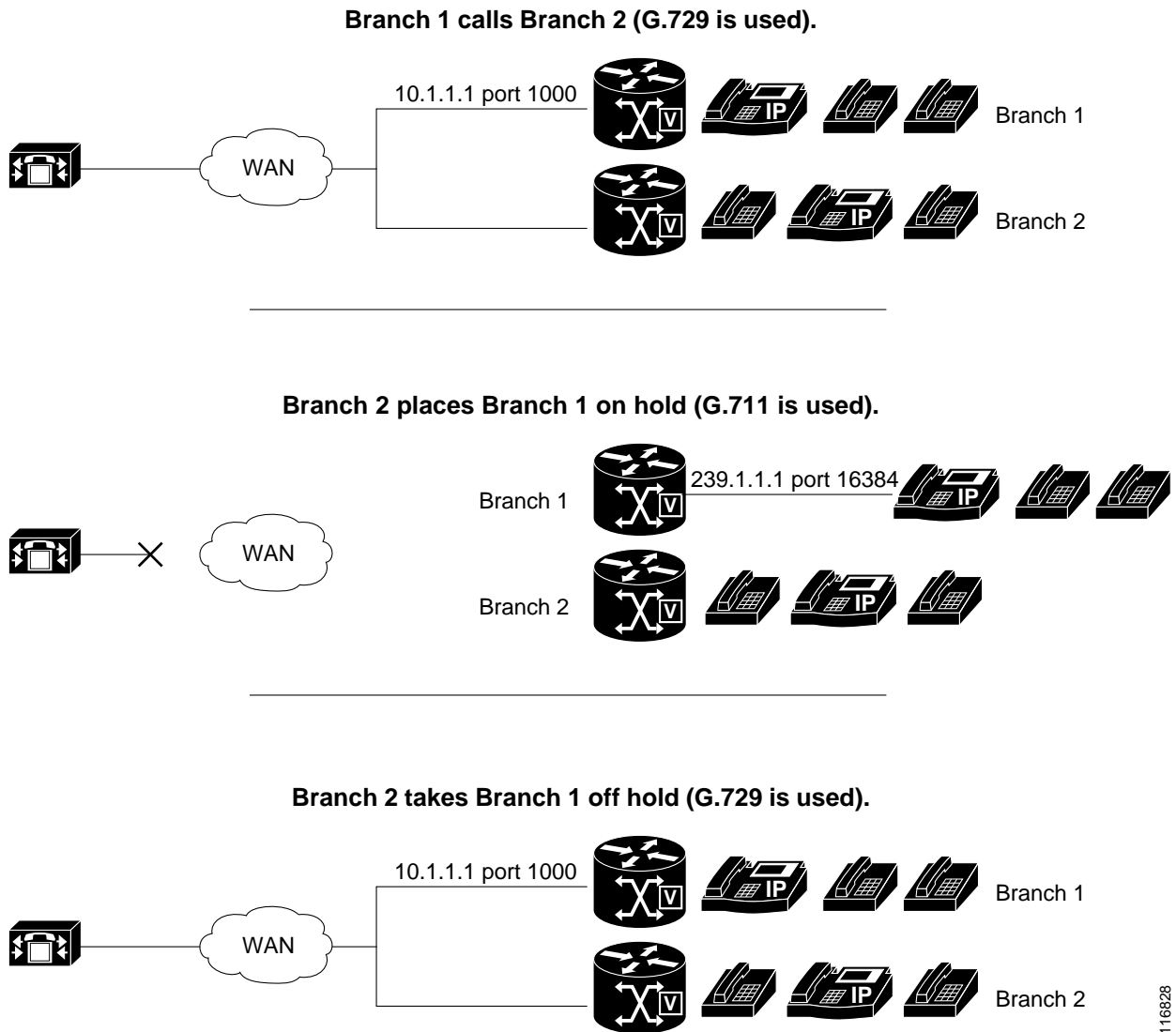


Codecs, Port Numbers, and IP Addresses

Cisco SRST multicast MOH supports G.711 only. Figure 2 shows an example in which G.711 is the only codec used by a central Cisco CallManager and three branches. In some cases, a Cisco CallManager system may use additional codecs. For example, for bandwidth savings, Cisco CallManager may use G.711 for multicast MOH and G.729 for phone conversations.

As shown in the example in Figure 2, IP address 10.1.1.1 and port 1000 are used during phone conversations when G.729 is in use, and IP address 239.1.1.1 and port 16384 are used when a call is placed on hold and G.711 is in use.

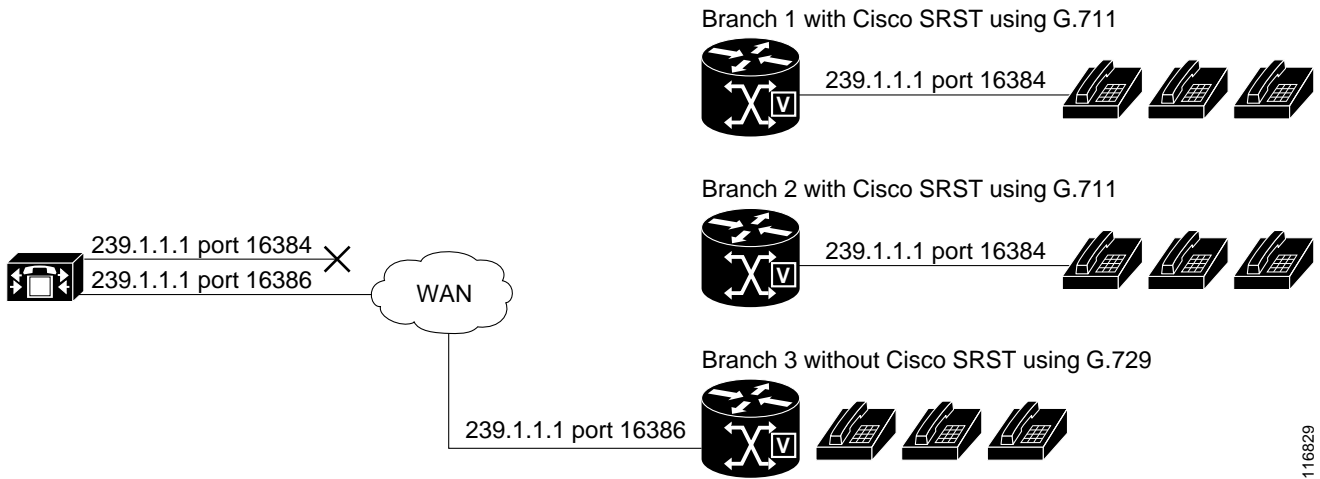
Figure 2 IP Address and Port Usage for G.711 and G.729 Configuration



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Figure 1 and Figure 2 show all branches using Cisco SRST multicasting MOH. Figure 3 shows a case in which some gateways are configured with Cisco SRST and other gateways are not. When the central site and Branch 3 phone users are put on hold by other IP phones in the Cisco CallManager system, MOH is originated by Cisco CallManager. When Branch 1 and Branch 2 phone users are put on hold by other phone users in the Cisco CallManager system, MOH is originated by the Cisco SRST gateways.

Figure 3 MOH Sources for Cisco SRST and Non-Cisco SRST IP Phones Using MOH



To enable MOH audio packet transmission through two paths, the Cisco CallManager MOH server must be configured with either one IP address and two different port numbers or one port address and two different IP multicast addresses so that one set of branches can use Cisco SRST multicast MOH and the other can use Cisco CallManager multicast MOH.

Multicast MOH Transmission

If Cisco SRST multicast MOH is supported by all branches in a system, such as in [Figure 1](#), Cisco CallManager must be configured to keep all multicast MOH audio packets from reaching the WAN. When there is a mix of Cisco SRST branches, as shown in [Figure 3](#), one set of Cisco CallManager MOH audio files must reach the WAN and another set must not. Audio packets from the central Cisco CallManager must cross the WAN to reach branches running Cisco CallManager. For branches running Cisco SRST, the packets must not reach the WAN.

SRST MOH Live Feed Support

Cisco SRST has been enhanced with the **moh-live** command. The **moh-live** command provides live feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. Music from a live feed is from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. Live feed MOH can also be multicast to Cisco IP phones. See the [“Configuring SRST MOH Live-Feed Support”](#) section on [page 25](#) for configuration instructions.

How to Use Cisco SRST Gateways as a Multicast MOH Resource

The three main tasks for configuring Cisco SRST gateways as an MOH resource are the following:

- [Configuring Cisco CallManager for Cisco SRST Multicast MOH, page 8](#) (required)
- [Configuring Cisco SRST for Multicast MOH, page 17](#) (required)
- [Configuring SRST MOH Live-Feed Support, page 25](#) (optional)

Configuring Cisco CallManager for Cisco SRST Multicast MOH

The following sections describe the Cisco CallManager configuration tasks for Cisco SRST multicast MOH:

- [Configuring the MOH Audio Source to Enable Multicasting, page 9](#)
- [Enabling Multicast on the Cisco CallManager MOH Server and Configuring Port Numbers and IP Addresses, page 10](#)
- [Creating an MRG and an MRGL, Enabling MOH Multicast, and Configuring Gateways, page 13](#)
- [Creating a Region for the MOH Server, page 15](#)
- [Verifying Cisco CallManager Multicast MOH, page 16](#)

To use Cisco SRST gateways as multicast MOH resources, you must configure Cisco CallManager to multicast MOH to the desired branch sites. To accomplish this, you must configure IP addresses, ports numbers, the MOH source, and the MOH audio server.

Configuring the MOH server involves designating a maximum number of hops for the audio source. A configuration of one hop keeps Cisco CallManager multicast MOH packets from reaching the WAN, thus spoofing Cisco CallManager and allowing Cisco SRST multicast MOH packets to be sent from Cisco SRST gateways to their component phones. For cases in which Cisco CallManager multicast must reach gateways that do not run Cisco SRST, use the Cisco IOS **ip multicast boundary** command to control where multicast packets go.

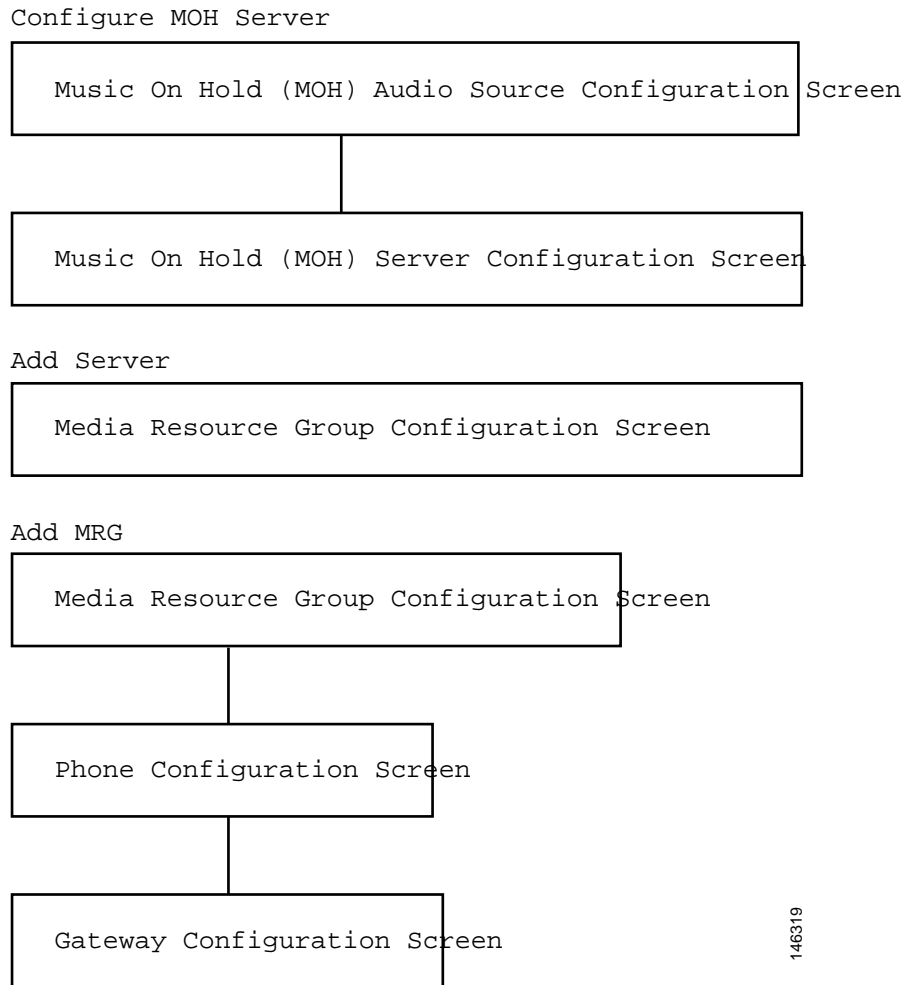
After the MOH server has been configured, the MOH server must be added to a Media Resource Group (MRG); the MRG is added to a Media Resource Group List (MRGL); and the designated Cisco CallManager branch gateways are configured to use the MRGL.

Five Cisco CallManager windows are used to configure the MOH server, audio source, MRG, MRGL, and individual gateways. [Figure 4](#) provides an overview of this process.

The last Cisco CallManager configuration task involves creating an MOH region that assigns MOH G.711 codec usage for the central site or sites and branch office or offices.

Regions specify the codecs that are used for audio and video calls within a region and between existing regions. For information about regions, see the “Region Configuration” section in the *Cisco CallManager Administration Guide*. From the [Cisco CallManager](#) documentation directory, click the desired Cisco CallManager version and go to **System Administration and Features and Services > Cisco CallManager Administration, System, and Features and Services Guides > Cisco CallManager Administration Guide > Region Configuration**.

Figure 4 CallManager Screens for Configuring Multicast MOH



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Configuring the MOH Audio Source to Enable Multicasting

The MOH audio source is a file from which Cisco CallManager transmits RTP packets. You can create an audio file or use the default audio file. For Cisco SRST multicast MOH, only one audio source can be used, even if, for example, one out of 500 sites uses Cisco SRST multicast MOH. In addition, all Cisco CallManager systems must use the same audio source for user and network MOH because Cisco SRST multicast MOH can stream audio only to a single multicast IP address and port. For Cisco SRST multicast MOH, the Cisco CallManager audio source file must be configured for G.711 bandwidth.



Tip The simplest way to create an audio source is to use the default audio source.

Whether you use a default Cisco CallManager MOH audio source or you create one, the MOH audio source must be configured for multicasting in the Music On Hold (MOH) Audio Source Configuration window.

Note that the MOH Audio Source File Status section shows that the MOH audio source file is configured for four codec formats. If you are planning to use several codecs, you must ensure that the audio source file accommodates them.

For further information about the creation of an MOH audio source, go to the [Cisco CallManager](#) documentation directory, click the desired Cisco CallManager version, and go to **System Administration and Features and Services > Cisco CallManager Administration, System, and Features and Services Guides > Cisco CallManager Features and Services Guide > Music On Hold > Creating Audio Sources**.

To configure the MOH audio source to enable multicasting and continuous play, perform the steps that follow.



Note

These instructions assume that an MOH audio source file has already been created.

SUMMARY STEPS

1. Enable multicast MOH for the MOH audio source.
2. Enable the audio source.
3. Allow multicasting.
4. Apply all multicasting changes.

DETAILED STEPS

-
- | | |
|---------------|--|
| Step 1 | To enable multicast MOH for the MOH audio source, choose Service > Media Resources > Music On Hold Audio Source to display the Music On Hold (MOH) Audio Source Configuration window. |
| Step 2 | Double-click the desired audio source listed in the MOH Audio Sources column. |
| Step 3 | In the Music On Hold (MOH) Audio Source Configuration window, check Allow Multicasting . |
| Step 4 | Click Update . |
-

Enabling Multicast on the Cisco CallManager MOH Server and Configuring Port Numbers and IP Addresses

You must enter a base multicast IP address and port number in the Multicast Audio Source Information section of the Music On Hold (MOH) Server Configuration window. If you are using Cisco CallManager multicast MOH and Cisco SRST multicast MOH (see the [“Codecs, Port Numbers, and IP Addresses”](#) section on page 5 and the [“Multicast MOH Transmission”](#) section on page 7), you must select a port and IP address increment method to configure for two sets of port numbers and IP address.

If the Increment Multicast on radio button is set to IP Address, each MOH audio source and codec combination is multicast to different IP addresses but uses the same port number. If it is set to Port Number, each MOH audio source and codec combination is multicast to the same IP address but uses different destination port numbers.

[Table 1](#) shows the difference between incrementing on an IP address and incrementing on a port number, using the base IP address of 239.1.1.1 and the base port number of 16384. The table also matches Cisco CallManager audio sources and codecs to IP addresses and port numbers.

Table 1 *Example of the Differences Between Incrementing Multicast on IP Address and Incrementing Multicast on Port Number*

| Audio Source | Codec | Increment Multicast on IP Address | | Increment Multicast on Port Number | |
|--------------|--------------|-----------------------------------|------------------|------------------------------------|------------------|
| | | Destination IP Address | Destination Port | Destination IP Address | Destination Port |
| 1 | G.711 mu-law | 239.1.1.1 | 16384 | 239.1.1.1 | 16384 |
| 1 | G.711 a-law | 239.1.1.2 | 16384 | 239.1.1.1 | 16385 |
| 1 | G.729 | 239.1.1.3 | 16384 | 239.1.1.1 | 16386 |
| 1 | Wideband | 239.1.1.4 | 16384 | 239.1.1.1 | 16387 |
| 2 | G.711 mu-law | 239.1.1.5 | 16384 | 239.1.1.1 | 16388 |
| 2 | G.711 a-law | 239.1.1.6 | 16384 | 239.1.1.1 | 16389 |
| 2 | G.729 | 239.1.1.7 | 16384 | 239.1.1.1 | 16390 |
| 2 | Wideband | 239.1.1.8 | 16384 | 239.1.1.1 | 16391 |

Incrementation is triggered by a change in codec usage. When codec usage changes, a new IP address or port number (depending on the incrementation selected) is assigned to the new codec type and is put into use. The original codec keeps its IP address and port number. For example, as seen in [Table 1](#), if your baseline IP address and port number are 239.1.1.1 and 16384 for a G.711 mu-law codec and the codec usage changes to G.729 (triggering an increment on the port number), the IP address and port number in use changes, or increment, to 239.1.1.1 and 16386. If G.711 usage resumes, the IP address and port number returns to 239.1.1.1 and 16384. If G.729 is in use again, the IP address and port goes back to 239.1.1.1 and 16386, and so forth.

It is important to configure a Cisco CallManager port number and IP address that use a G.711 audio source for Cisco SRST multicast MOH. If Cisco CallManager multicast MOH is also being used on gateways that do not have Cisco SRST and use a different codec, such as G.729, ensure that the additional or incremental port number or IP address uses the same audio source as the Cisco SRST gateways and the required codec.

The Music On Hold (MOH) Server Configuration window is also where the multicast audio source for the MOH server is configured. For Cisco SRST multicast MOH, the Cisco CallManager MOH server can use only one audio source. An audio source is selected by inputting the audio source's maximum number of hops.

The Max Hops configuration sets the length of the transmission of the audio source packets. Limiting the number of hops is one way to stop audio packets from reaching the WAN and thus spoofing Cisco CallManager so Cisco SRST can multicast MOH. If all of your branches run Cisco SRST, use a low number of hops to prevent audio source packets from crossing the WAN. If your system configuration includes routers that do not run Cisco SRST, enter a high number of hops to allow source packets to cross the WAN. Use the **ip multicast borderer** and **access-list** commands to keep resource packets from specific IP addresses from reaching the WAN.

To enable multicast and configure port numbers and IP addresses, perform the following steps.

SUMMARY STEPS

1. Enable multicast MOH for Cisco CallManager.
2. Set the base IP address and port number.
3. Select whether Cisco CallManager increments port numbers or IP addresses.
4. Enter a maximum number of hops.
5. Use Cisco IOS commands to stop Cisco CallManager signals from crossing the WAN and reaching Cisco SRST gateways.

DETAILED STEPS

Step 1 Enable multicast MOH for Cisco CallManager.

- a. Choose **Service > Media Resource > Music On Hold Server**.
- b. The Music On Hold (MOH) Server Configuration window appears.
- c. Call up an existing MOH server by clicking **Find** and double-clicking the desired MOH server.
- d. Regardless of whether you are updating or creating an MOH server, click **Enable Multicast Audio Sources on this MOH Server**.

Step 2 Set the base IP address and port number.

In the Music On Hold (MOH) Server Configuration window, enter an IP address in the Base Multicast IP Address field and enter a port number in the Base Multicast Port Number field. Be sure that the IP address and port number use the desired audio source and codec. See [Table 1](#).

Step 3 Select whether Cisco CallManager increments port numbers or IP addresses.

In the Music On Hold (MOH) Server Configuration window, in the Increment Multicast on field, choose **Port Number** if you want port numbers to be incremented and the IP address to remain unchanged. Choose **IP Address** if you want IP addresses to be incremented and the port number to remain unchanged.

- If all of your branches run Cisco SRST and thus use G.711 for MOH, use either setting because incrementation does not take place and a selection does not matter.
- If your system configuration includes routers that do not run Cisco SRST and use a different codec, select an incrementation method.



Note If your branches include routers that do not run Cisco SRST but do use G.711, configure separate audio sources: one for the routers that run Cisco SRST and one for the routers that do not.

Step 4 Enter a maximum number of hops.

In the Music On Hold (MOH) Server Configuration window, next to the Audio Source Name field, enter 1 in the Max Hops field if all of your branches run Cisco SRST. If your system configuration includes routers that do not run Cisco SRST, enter 16 in the Max Hops field.

Step 5 Use Cisco IOS commands to stop Cisco CallManager signals from crossing the WAN and reaching Cisco SRST gateways.

If all of your branches run Cisco SRST, skip this step. If your system configuration includes routers that do not run Cisco SRST and use a different codec, enter the following Cisco IOS commands starting from global configuration mode on the central site router:

a. **interface** {**serial** | **fastethernet**} *slot/port*

Enters interface configuration mode, where *slot* is the slot number and *port* is the port number.

b. **ip multicast boundary** *access-list-number*

Configures an administratively scoped boundary, where *access-list-number* is a number from 1 to 99 that identifies an access list that controls the range of group addresses affected by the boundary.

c. **exit**

Exits interface configuration mode.

d. **access-list** *access-list-number* **deny** *ip-address*

Configures the access list mechanism for filtering frames by IP address. For the *ip-address* argument, enter the MOH IP address that you want to prevent from going over the WAN. Normally this would be the base IP address entered in Step 2.

The following is an example configuration:

```
Router(config)# interface serial 0/0
Router(config-if)# ip multicast boundary 1
Router(config-if)# exit
Router(config)# access-list 1 deny 239.1.1.1
```

Creating an MRG and an MRGL, Enabling MOH Multicast, and Configuring Gateways

The next task involves configuring individual gateways to use an MOH server that can transport the desired MOH audio source to their IP phones on hold. This is accomplished by creating a Media Resource Group (MRG). An MRG references media resources, such as MOH servers. The MRG is then added to a Media Resource Group List (MRGL), and the MRGL is added to the phone and gateway configurations.

MRGs are created in the Media Resource Group Configuration window. MRGLs are created in the Media Resource Group List Configuration window. Phones are configured in the Phone Configuration window. Gateways are configured in the Gateway Configuration window.



Note

The Gateway Configuration window for an H.323 gateway is similar for MGCP gateways.

An MRGL can also be added to a gateway or IP phone configuration by adding the MRGL to a device pool configuration. For further information about device pools, go to the [Cisco CallManager](#) documentation directory, click the desired Cisco CallManager version, and go to **System Administration and Features and Services > Cisco CallManager Administration, System, and Features and Services Guides > Cisco CallManager Administration Guide > Device Pool Configuration**.

To create an MRG and MRGL, to enable MOH multicast, and to configure gateways, perform the following steps.

SUMMARY STEPS

1. Create an MRG with a multicast MOH media resource.
2. Create an MRGL that contains the newly created MRG.
3. Add the MRGL to the desired IP phones.
4. Add the MRGL to the desired gateway.

DETAILED STEPS

Step 1 Create an MRG with a multicast MOH media resource.

- a. Choose **Service > Media Resource > Media Resource Group**.
- b. In the upper-right corner of the window, click the **Add a New Media Resource Group** link. The Media Resource Group Configuration window appears.
- c. Complete the Media Resource Group Name field.
- d. Complete the Description field.
- e. Select a media resource from the Available Media Resources pane.

This pane lists the media resources that can be chosen for an MRG and can include the following media resource types:

- Conference bridges (CFB)
- Media termination points (MTP)
- Music-on-hold servers (MOH)
- Transcoders (XCODE)
- Annunciator (ANN)

Music-on-hold servers that are configured for multicast are labeled as (MOH) [Multicast].

- f. Click the down arrow so that the selected media resource moves to the Selected Media Resources pane.
- g. Click **Insert**.

Step 2 Create an MRGL that contains the newly created MRG.

- a. Choose **Service > Media Resource > Media Resource Group List**.
- b. In the upper-right corner of the window, click the **Add a New Media Resource Group List** link. The Media Resource Group List Configuration window appears.
- c. Complete the Media Resource Group List Name field.
- d. In the Available Media Resource Groups pane, select the MRG that you just created.
- e. Add the MRG to the Selected Media Resource Groups pane by clicking the down arrow. After a media resource group is added, its name moves to the Selected Media Resource Groups pane.
- f. Click **Insert**.

Step 3 Add the MRGL to the desired IP phones.

- a. Choose **Device > Phone** to display the Find and List Phones window.

- b. Click **Find** to display a list of phones.
- c. Double-click the device name of the phone that you want to update.
- d. Complete the Media Resource Group List field by choosing the desired MRGL from the drop-down menu.
- e. Click **Update**.

Step 4 Add the MRGL to the desired gateway.

- a. Choose **Device > Gateway** to display the Find and List Gateways window.
 - b. Click **Find** to display a list of gateways.
 - c. Double-click the device name of the gateway that you want to update.
 - d. If the gateway is H.323, complete the Media Resource Group List field by choosing the desired MRGL from the drop-down menu.
 - e. Click **Update**.
-

Creating a Region for the MOH Server

To ensure that the MOH server uses G.711 for Cisco SRST gateways, you must create a separate region for the MOH server. For more information about codecs, see the [“Codecs, Port Numbers, and IP Addresses” section on page 5](#). For information about regions, go to the [Cisco CallManager documentation directory](#), click the desired Cisco CallManager version and go to **System Administration and Features and Services > Cisco CallManager Administration, System, and Features and Services Guides > Cisco CallManager Administration Guide > Region Configuration**.

Regions are configured in the Region Configuration window. If the Cisco CallManager system uses G.711 only, all of the central sites and their constituent branches for the MOH region must be set to G.711. If a Cisco CallManager system has a combination of branches that do and do not run Cisco SRST multicast MOH and the branches that do not run Cisco SRST require a different codec for Cisco CallManager multicast MOH, they must be configured accordingly.

A Region Configuration window where the “MoH Server” region is configured to use the G.711 and G.729 codecs might look like this:

- G.711 is used for Branch 1 because its gateway is configured to run Cisco SRST multicast MOH, which requires G.711.
- G.729 is used for Branch 2 because its gateway does not run Cisco SRST and it is configured to use a port and IP address that use G.729.
- G.711 is configured for the central site and the MOH server region.

To create a region for the MOH server, perform the following steps.

SUMMARY STEPS

1. Create an MOH server region.
2. Create other regions as needed for different codecs.

DETAILED STEPS

Step 1 Create an MOH server region.

- a. Choose **System > Region**.

- b. In the upper-right corner of the window, click **Add a New Region**. The Region Configuration window appears.
- c. In the Region Name field, enter the name that you want to assign to the new region and click **Insert**.
- d. If other regions have been created, a list of regions appear. Use the drop-down list boxes to choose the audio codec to use for calls between the new region and existing regions. The audio codec determines the type of compression and the maximum amount of bandwidth that is allocated for these calls.
- e. In addition to other regions, the newly created region appears in the list. Use its drop-down box to choose the codec for use within the new region.
- f. Click **Update**.

Step 2 Create other regions as needed for different codecs.

Verifying Cisco CallManager Multicast MOH

The Cisco CallManager multicast MOH configuration must run correctly in order for Cisco SRST multicast MOH to work. Verification of Cisco CallManager multicast MOH differs for configurations using a WAN with multicast enabled and a WAN with multicast disabled.

It is important to verify that the Cisco CallManager multicast MOH is provided through multicasting and not unicasting. Because unicast MOH is enabled by default, it is easy to mistakenly conclude that multicast MOH is working when it is not.

SUMMARY STEPS

1. Verify that Cisco CallManager's multicast MOH is heard on a remote gateway.
2. Verify that Cisco CallManager's MOH is multicast, not unicast.

DETAILED STEPS

Step 1 Verify that Cisco CallManager's multicast MOH is heard on a remote gateway.

- a. If multicast is enabled on the WAN, make sure that the number of hops configured on the Cisco CallManager MOH server is sufficient to allow audio packets to reach the remote site (see the [“Enabling Multicast on the Cisco CallManager MOH Server and Configuring Port Numbers and IP Addresses”](#) section on page 10). Then call an IP phone on a remote gateway, place the call on hold, and verify that MOH is heard.
- b. If multicast is not enabled on the WAN, place an IP phone on the same subnet as the Cisco CallManager MOH server and verify that MOH can be heard. Because the IP phone and the MOH server are on the same subnet, no multicast routing capabilities in the network are required.

Step 2 Verify that the Cisco CallManager's MOH is multicast, not unicast.

- a. From Microsoft Windows, select **Start > Programs > Administrative Tools > Performance**.
- b. In the Performance window, click the + (plus) icon located at the top of the right pane.
- c. In the Add Counters window, select Cisco MOH Device.
- d. In the Performance window, you can monitor the MOHMulticastResourceActive and MOHUnicastResourceActive counters to check on multicast activity.

Configuring Cisco SRST for Multicast MOH

Cisco SRST configuration includes the following:

- [Enabling Multicast MOH on the Cisco SRST Gateway, page 17](#)
- [Verifying Basic Cisco SRST Multicast MOH Streaming, page 20](#)
- [Verifying Cisco SRST MOH to PSTN, page 21](#)
- [Verifying Cisco SRST Multicast MOH to IP Phones, page 24](#)

Prerequisites

- The SRST gateways must run Cisco IOS Release 12.2(15)ZJ2 or later.
- The flash memory in each of the Cisco SRST gateways must have an MOH audio file. The MOH file can be in .wav or .au file format, but must contain 8-bit 8-kHz data, such as an a-law or mu-law data format. A known working MOH audio file (music-on-hold.au) is included in the program .zip files that can be downloaded from <http://www.cisco.com/cgi-bin/tablebuild.pl/ip-key>. Or the music-on-hold.au file can be downloaded from <http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp> and copied to the flash memory on your Cisco SRST router. Another option is copying one of the MOH files, such as SampleAudioSource.ULAW.wav, from C:\Program Files\Cisco\MOH on Cisco CallManager.



Note During the copying process, four files are added to each router's flash automatically. One of the files must use a mu-law format as indicated by the extension .ULAW.wav.

- A loopback interface must be configured, and its IP addresses must be included in the Cisco SRST multicast MOH configuration. This configuration allows multicast MOH to be heard on POTS ports on the gateway. The loopback interface does not have to bind to either H.323 or MGCP.
- At least one ephone and directory number (DN) must be configured, even if the gateway is not used for Cisco SRST. Cisco SRST multicast MOH streaming never starts without an ephone and directory number.

Enabling Multicast MOH on the Cisco SRST Gateway

No multicast MOH routing configuration is required for Cisco SRST gateways because each SRST gateway is configured to act as a host running an application that streams multicast MOH packets from the network. The **multicast moh** command declares the Cisco CallManager multicast MOH address and port number and allows Cisco SRST gateways to route MOH from flash memory to up to four IP addresses. If no route IP addresses are configured, the flash MOH is sent through the IP address configured in the Cisco SRST **ip source-address** command.

SUMMARY STEPS

1. **ccm-manager music-on-hold**
2. **interface loopback** *number*
3. **ip address** *ip-address mask*
4. **exit**
5. **interface fastethernet** *slot/port*
6. **ip address** *ip-address mask*
7. **exit**
8. **call-manager-fallback**
9. **ip source-address** *ip-address* [**port** *port*]
10. **max-ephones** *max-phones*
11. **max-dn** *max-directory-number*
12. **moh** *filename*
13. **multicast moh** *multicast-address* **port** *port* [**route** *ip-address-list*]
14. **exit**

DETAILED STEPS

| | Command or Action | Purpose |
|--------|---|---|
| Step 1 | ccm-manager music-on-hold Example: Router(config)# ccm-manager music-on-hold | Enables the multicast MOH feature on a voice gateway. |
| Step 2 | interface loopback <i>number</i> Example: Router(config)# interface loopback 1 | Configures an interface type and enters the interface configuration mode. <ul style="list-style-type: none"> • <i>number</i>—Loopback interface number. The range is from 0 to 2147483647. |
| Step 3 | ip address <i>ip-address mask</i> Example: Router(config-if)# ip address 10.1.1.1 255.255.255.255 | Sets a primary IP address for an interface. <ul style="list-style-type: none"> • <i>ip-address</i>—IP address. • <i>mask</i>—Mask for the associated IP subnet. |
| Step 4 | exit Example: Router(config-if)# exit | Exits interface configuration mode. |
| Step 5 | interface fastethernet <i>slot/port</i> Example: Router(config)# interface fastethernet 0/0 | (Optional if the route keyword is not used in the multicast moh command. See Step 9 and Step 13.) Configures an interface type and enters interface configuration mode. |

| | Command or Action | Purpose |
|---------|--|---|
| Step 6 | <p>ip address <i>ip-address mask</i></p> <p>Example: Router(config-if)# ip-address 172.21.51.143 255.255.255.192</p> | (Optional if the route keyword is not used in the multicast moh command. See Step 9 and Step 13.) Sets a primary IP address for an interface. |
| Step 7 | <p>exit</p> <p>Example: Router(config-if)# exit</p> | (Optional if the route keyword is not used in the multicast moh command. See Step 9 and Step 13.) Exits interface configuration mode. |
| Step 8 | <p>call-manager-fallback</p> <p>Example: Router(config)# call-manager-fallback</p> | Enters call-manager-fallback configuration mode. |
| Step 9 | <p>ip source-address <i>ip-address [port port]</i></p> <p>Example: Router(config-cm-fallback)# ip source-address 172.21.51.143 port 2000</p> | <p>(Optional if the route keyword is not used in the multicast moh command. See Step 13.) Enables a router to receive messages from Cisco IP phones through the specified IP addresses and ports.</p> <ul style="list-style-type: none"> • <i>ip-address</i>—The preexisting router IP address, typically one of the addresses of the Ethernet port of the router. • port port—(Optional) The port to which the gateway router connects to receive messages from the Cisco IP phones. The port number range is from 2000 to 9999. The default port number is 2000. |
| Step 10 | <p>max-ephones <i>max-phones</i></p> <p>Example: Router(config-cm-fallback)# max-ephones 1</p> | <p>Configures the maximum number of Cisco IP phones that can be supported by a router.</p> <ul style="list-style-type: none"> • <i>max-phones</i>—Maximum number of Cisco IP phones supported by the router. The maximum number is platform-dependent. The default is 0. |
| Step 11 | <p>max-dn <i>max-directory-number</i></p> <p>Example: Router(config-cm-fallback)# max-dn 1</p> | <p>Sets the maximum possible number of virtual voice ports that can be supported by a router.</p> <ul style="list-style-type: none"> • <i>max-directory-number</i>—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum possible number is platform-dependent. The default is 0. |
| Step 12 | <p>moh <i>filename</i></p> <p>Example: Router(config-cm-falback)# moh music-on-hold.au</p> | <p>Enables use of an MOH file.</p> <ul style="list-style-type: none"> • <i>filename</i>—Filename of the music file. The music file must reside in flash memory. |

| Command or Action | Purpose |
|---|--|
| <p data-bbox="142 331 224 361">Step 13</p> <pre data-bbox="240 331 860 386">multicast moh <i>multicast-address</i> port <i>port</i> [route <i>ip-address-list</i>]</pre> <p data-bbox="240 436 344 466">Example:</p> <pre data-bbox="240 470 932 550">Router(config-cm-fallback)# multicast moh 10.10.255.250 port 2 route 10.10.255.251 10.10.255.252 10.10.255.253 10.10.255.254</pre> | <p data-bbox="954 331 1521 394">Enables multicast of MOH from a branch office flash MOH file to IP phones in the branch office.</p> <ul data-bbox="971 415 1521 1150" style="list-style-type: none"> <li data-bbox="971 415 1521 886">• multicast-address and port port—Declares the IP address and port number of MOH packets that are to be multicast. The multicast IP address and port must match the IP address and the port number that Cisco CallManager is configured to use for multicast MOH. If you are using different codecs for MOH, these may not be the base IP address and port, but an incremented IP address or port number. See the “Configuring the MOH Audio Source to Enable Multicasting” section on page 9. If you have multiple audio sources configured on Cisco CallManager, ensure that you are using the audio sources’s correct IP address and port number. <li data-bbox="971 907 1521 1150">• route ip-address-list—(Optional) Declares the IP address or addresses from which the flash MOH packets can be transmitted. A maximum of four IP address entries are allowed. If a route keyword is not configured, the Cisco SRST system uses the ip source-address command value configured for Cisco SRST. |
| <p data-bbox="142 1171 224 1201">Step 14</p> <pre data-bbox="240 1171 298 1201">exit</pre> <p data-bbox="240 1251 344 1281">Example:</p> <pre data-bbox="240 1285 656 1306">Router(config-cm-fallback)# exit</pre> | <p data-bbox="954 1171 1521 1201">Exits call-manager-fallback configuration mode.</p> |

Verifying Basic Cisco SRST Multicast MOH Streaming

To verify that multicast MOH packets are configured with the **multicast moh** command, perform the following steps.

SUMMARY STEPS

1. **debug ephone moh**
2. **show interfaces fastethernet**
3. **show ephone summary**

DETAILED STEPS

Step 1 **debug ephone moh**

This command sets debugging for MOH. You can use this command to show that the Cisco SRST gateway is multicasting MOH out of Loopback 0 and Fast Ethernet 0/0:

```
Router# debug ephone moh
!
MOH route If FastEthernet0/0 ETHERNET 172.21.51.143 via ARP
MOH route If Loopback0 46 172.21.51.98 via 172.21.51.98
!
```

Step 2 **show interfaces fastethernet**

Use this command to confirm that the interface output rates match one G.711 stream, which the **show interfaces fastethernet** output displays as 50 packets/sec and 80 kbps or more.

```
Router# show interfaces fastethernet 0/0
!
30 second output rate 86000 bits/sec, 50 packets/sec
!
```

Step 3 **show ephone summary**

Use this command to verify that the Cisco IOS software was able to read the MOH audio file successfully.

```
Router# show ephone summary
!
File music-on-hold.au type AU Media_Payload_G.711Ulaw64k 160 bytes
!
```

Troubleshooting Tips

The **show ephone summary** output should show a file type as either .au or .wav. If INVALID appears, an error exists.

```
Router# show ephone summary
!
File music-on-hold.au type INVALID Media_Payload_G.711Ulaw64k 160 bytes651-
!
```

An invalid output may be caused by the order in which the Cisco SRST configuration commands are entered. Issue the **no call-manager-fallback** command and reenter the multicast MOH commands. Rebooting may also clear the error.

Verifying Cisco SRST MOH to PSTN

To verify that multicast MOH packets transmit over a public switched telephone network (PSTN), perform the following steps.

SUMMARY STEPS

1. Verify that a PSTN caller hears MOH when placed on hold by an IP phone caller.
2. **show ccm-manager music-on-hold**
3. **debug h245 asn**
4. **show call active voice**

DETAILED STEPS

Step 1 Verify that a PSTN caller hears MOH when placed on hold by an IP phone caller.

Use a Cisco SRST gateway IP phone to call a PSTN phone, and put the PSTN caller on hold. The PSTN caller should hear MOH.

Step 2 **show ccm-manager music-on-hold**

Use this command to verify that the MOH is multicast. Note that the **show ccm-manager music-on-hold** command displays information about PSTN connections on hold only. It does not display information about multicast streams going to IP phones on hold. The following is an example of **show ccm-manager music-on-hold** command output.

```
Router# show ccm-manager music-on-hold
```

```
Current active multicast sessions : 1
Multicast      RTP port  Packets      Call   Codec   Incoming
Address        number    in/out       id     G.711ulaw  Interface
=====
239.1.1.1      16384    326/326     42    G.711ulaw  Lo0
```

If the PSTN caller hears MOH, but the **show ccm-manager music-on-hold** command displays no active multicast streams, the MOH is unicast. This can be confirmed by checking the MOH performance counters as discussed in the [“Verifying Cisco CallManager Multicast MOH”](#) section on page 16.

Step 3 **debug h245 asn**

Use this command if H.323 is being used and no multicast address appears in the **show ccm-manager music-on-hold** command output to verify the H.323 handshaking between Cisco CallManager and the Cisco SRST gateway. When a PSTN caller is placed on hold, Cisco CallManager sends an H.245 closeLogicalChannel, followed by an openLogicalChannel. Verify that the final openLogicalChannelAck from Cisco CallManager to the Cisco SRST gateway contains the expected multicast IP address and port number. In the following example, the IP address is EF010101 (239.1.1.1) and the port number is 16384.

```
Router# debug h245 asn
```

```
*Mar  1 04:20:19.227: H245 MSC INCOMING PDU ::=
value MultimediaSystemControlMessage ::= response : openLogicalChannelAck :
{
  forwardLogicalChannelNumber 6
  forwardMultiplexAckParameters h2250LogicalChannelAckParameters :
  {
    sessionID 1
    mediaChannel unicastAddress : ipAddress :
    {
      network 'EF010101'H
      tsapIdentifier 16384
    }
    mediaControlChannel unicastAddress : ipAddress :
    {
      network 'EF010101'H
      tsapIdentifier 16385
    }
  }
}
```

Step 4 show call active voice

Use this command with the **debug h245 asn** command to further verify the H.323 handshaking between Cisco CallManager and the Cisco SRST gateway.

```
Router# show call active voice | include RemoteMedia
```

```
RemoteMediaIPAddress=239.1.1.1  
RemoteMediaPort=16384
```

The IP address and port number displayed should match the IP address and port number displayed by the **debug h245 asn** command. If the RemoteMediaIPAddress field displays 0.0.0.0, it is likely that you have encountered caveat CSCdz00697. For more information, see the [Bug ToolKit](#) and the [“Restrictions for Using Cisco SRST Gateways as a Multicast MOH Resource”](#) section on page 3.

Troubleshooting Tips

- If the PSTN caller hears tone on hold (TOH) instead of MOH, two problems are likely:
 - Cisco CallManager has failed to activate MOH and has used TOH as a fallback. To verify that this is the case, see the [“Verifying Cisco CallManager Multicast MOH”](#) section on page 16.
 - Cisco CallManager does not have the appropriate MOH resource available. Use the **show ccm-manager music-on-hold** command to find out if the MOH resource is the problem.



Note The **show ccm-manager music-on-hold** command displays information about PSTN connections on hold only. It does not display information about multicast streams going to IP phones on hold.

```
Router# show ccm-manager music-on-hold
```

```
Current active multicast sessions : 1  
Multicast      RTP port  Packets      Call   Codec   Incoming  
Address        number    in/out       id     id      Interface  
=====
```

| | | | | | |
|-----------|-------|---------|----|-----------|------|
| 239.1.1.1 | 16384 | 326/326 | 42 | G.711ulaw | Lo0* |
|-----------|-------|---------|----|-----------|------|

If no MOH streams are shown (that is, there are no rows of data beneath the columns), Cisco CallManager has not been correctly configured to provide the Cisco SRST gateway with MOH. Configuration errors may include that the required codec has not been enabled on Cisco CallManager (check the service parameters) and that no MRGL has been assigned to the gateway, or, if one has been assigned, it has insufficient resources. Check Cisco Intrusion Detection System (IDS) Event Viewer for error messages.

- If the POTS caller on hold does not hear a sound, Cisco CallManager has successfully completed the multicast MOH handshaking with the Cisco SRST gateway, but the gateway is failing to pick up the locally generated multicast RTP stream.

Use the **show ccm-manager music-on-hold** command to investigate.

```
Router# show ccm-manager music-on-hold
```

```
Current active multicast sessions : 1  
Multicast      RTP port  Packets      Call   Codec   Incoming  
Address        number    in/out       id     id      Interface  
=====
```

| | | | | | |
|-----------|-------|---------|----|-----------|-------|
| 239.1.1.1 | 16384 | 326/326 | 42 | G.711ulaw | Lo0 * |
|-----------|-------|---------|----|-----------|-------|

- If no MOH streams are shown, Cisco CallManager has not been correctly set up to provide the Cisco SRST gateway with MOH. A typical error is that Cisco CallManager has not been configured with an appropriate MOH resource. The configuration error might be that the required codec has not been enabled on Cisco CallManager (check the service parameters) or that no MRGL has been assigned to the gateway, or, if one has been assigned, it has insufficient resources. Check the IDS Event Viewer for error messages.
- Verify that the multicast address and RTP port number shown in the **show ccm-manager music-on-hold** command output match the *multicast-address* and *port* arguments in the **moh multicast** command configuration.
- Verify that the Packets in/out field shows a count that is incrementing. Repeat the **show ccm-manager music-on-hold** command to verify that the Packets in/out counters are incrementing.
- Verify that the codec field matches the codec type of the audio file stored in the Cisco SRST gateway’s flash memory. If another codec value besides G.711 mu-law or G.711 a-law appears in the **show ccm-manager music-on-hold** command output, review the Cisco CallManager region for incorrect codec configuration. See the [“Creating a Region for the MOH Server” section on page 15](#).
- The Incoming Interface field shows where the Cisco SRST gateway is to receive the multicast MOH packets. An interface must be listed, and it must be one of the interfaces included in the **multicast moh** command or the default IP source address, which is configured with the **ip source-address** command. See [Step 9](#) in the [“Enabling Multicast MOH on the Cisco SRST Gateway” section on page 17](#).

Verifying Cisco SRST Multicast MOH to IP Phones

To verify that Cisco CallManager is signaling the IP phone to receive Cisco SRST multicast MOH correctly, perform the following steps.

SUMMARY STEPS

1. Verify that an IP phone caller hears MOH when placed on hold by an IP phone caller.
2. Check the MOHMulticastResourceActive and MOHUnicastResourceActive counters.

DETAILED STEPS

Step 1 Verify that an IP phone caller hears MOH when placed on hold by an IP phone caller.

Use an IP phone to call a second IP phone, and put the second caller on hold. The second caller should hear MOH.

Step 2 Check the MOHMulticastResourceActive and MOHUnicastResourceActive counters.

Use the Performance window to check the MOHMulticastResourceActive and MOHUnicastResourceActive counters under the Cisco MOH Device performance object. See [Step 2](#) in the [“Verifying Cisco CallManager Multicast MOH” section on page 16](#). For Cisco SRST multicasting MOH to work, the multicast counter must increment.

Troubleshooting Tips

If no MOH is heard and the Cisco SRST MOH signaling is multicasting, connect a sniffer to the PC port on the back of IP phone. If the IP phone and Cisco SRST gateway are connected to the same subnet, multicast RTP packets should be detected at all times, even when the IP phone has not been placed on hold. If the IP phone and the Cisco SRST gateway are not connected to the same subnet, multicast RTP packets are detected only when the IP phone is placed on hold and sends an Internet Group Management Protocol (IGMP) Join to the closest router.

Configuring SRST MOH Live-Feed Support

The following sections describe the configuration tasks for Cisco SRST MOH live feed:

- [Setting Up the Voice Port on the Cisco SRST Gateway, page 26](#)
- [Setting Up the Directory Numbers on the Cisco SRST Gateway, page 27](#)
- [Establishing the MOH Feed, page 28](#)
- [Verifying Cisco SRST MOH Live Feed, page 30](#)

To configure MOH from a live feed, you first establish a voice port and dial peer for the call and then create a “dummy” phone or directory number. The dummy number allows for making and receiving calls, but the number is not assigned to a physical phone. It is that number that the MOH system autodials to establish the MOH feed.

The **moh-live** command allocates one of the virtual voice ports from the pool of virtual voice ports created by the **max-dn** command. The virtual voice port places an outgoing call to the dummy number; that is, the directory number specified in the **moh-live** command. The audio stream obtained from the MOH call provides the music-on-hold audio stream.

The recommended interface for live-feed MOH is an analog E&M port because it requires the minimum number of external components. You connect a line-level audio feed (standard audio jack) directly to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (An audio connection on an E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by a digital signal processor (DSP) on the E&M port.

If you are using an FXO voice port for live-feed MOH instead of an E&M port, connect the MOH source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip and ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip and ring leads of the FXO port.

Music from a live feed is continuously fed into the MOH playout buffer instead of being read from a flash file, so there is typically a 2-second delay. An outbound call to an MOH live-feed source is attempted (or reattempted) every 30 seconds until the connection is made by the directory number that has been configured for MOH. If the live-feed source is shut down for any reason, the flash memory source automatically activates.

A live-feed MOH connection is established as an automatically connected voice call that is made by the Cisco SRST MOH system itself or by an external source directly calling in to the live-feed MOH port. An MOH call can be from or to the PSTN or can proceed via VoIP with voice activity detection (VAD) disabled. The call is assumed to be an incoming call unless the **out-call** keyword is used with the **moh-live** command during configuration.

The Cisco SRST router uses the audio stream from the call as the source for the MOH stream, displacing any audio stream that is available from a flash file. An example of an MOH stream received over an incoming call is an external H.323-based server device that calls the directory number to deliver an audio stream to the Cisco SRST router.

Prerequisites

Cisco SRST for multicast MOH, as described in [“Configuring Cisco SRST for Multicast MOH” section on page 17](#), is not required for the MOH live-feed configuration. However, MOH live feed is designed to work in conjunction with multicast MOH.

Restrictions

- An FXO port can be used for a live feed if the port is supplied with an external third-party adapter to provide a battery feed.

- An FXS port cannot be used for a live feed.
- For a live feed from VoIP, VAD must be disabled.
- MOH is supplied only to PSTN and VoIP G.711 calls. Local IP phone callers hear a repeating tone on hold for reassurance that they are still connected.

Setting Up the Voice Port on the Cisco SRST Gateway

To activate MOH from a live feed, set up and connect the physical voice port.

SUMMARY STEPS

1. **voice-port** *port*
2. **input gain** *decibels*
3. **auto-cut-through** (E&M only)
4. **operation 4-wire** (E&M only)
5. **signal immediate** (E&M only)
6. **no shutdown**
7. **exit**

DETAILED STEPS

| | Command or Action | Purpose |
|--------|---|--|
| Step 1 | voice-port <i>port</i> Example: Router(config)# voice-port 1/1/0 | Enters voice-port configuration mode to set up the physical voice port. To find the correct definition of the <i>port</i> argument for your router, see the Cisco IOS Survivable Remote Site Telephony Version 3.2 Command Reference . |
| Step 2 | input gain <i>decibels</i> Example: Router(config-voice-port)# input gain 0 | Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable values are integers from -6 to 14. |
| Step 3 | auto-cut-through Example: Router(config-voiceport)# auto-cut-through | (E&M ports only) Enables call completion when a PBX does not provide an M-lead response. MOH requires that you use this command with E&M ports. |
| Step 4 | operation 4-wire Example: Router(config-voiceport)# operation 4-wire | (E&M ports only) Selects the 4-wire cabling scheme. MOH requires that you specify 4-wire operation with this command for E&M ports. |
| Step 5 | signal immediate Example: Router(config-voiceport)# signal immediate | (E&M ports only) For E&M tie trunk interfaces, directs the calling side to seize a line by going off-hook on its E-lead and to send address information as DTMF digits. |

| | Command or Action | Purpose |
|--------|--|--------------------------------------|
| Step 6 | <code>no shutdown</code> Example: Router(config-voiceport)# no shutdown | Activates the voice port. |
| Step 7 | <code>exit</code> Example: Router(config-voiceport)# exit | Exits voice-port configuration mode. |

Setting Up the Directory Numbers on the Cisco SRST Gateway

After setting up the voice port, create a dial peer and give the voice port a directory number with the **destination-pattern** command. The directory number is the number that the system uses to access the MOH.

SUMMARY STEPS

1. **dial-peer voice** *tag pots*
2. **destination-pattern** *string*
3. **port** *port*
4. **exit**

DETAILED STEPS

| | Command or Action | Purpose |
|--------|--|--|
| Step 1 | <code>dial-peer voice tag pots</code> Example: Router(config)# dial-peer voice 7777 pots | Enters dial-peer configuration mode. |
| Step 2 | <code>destination-pattern string</code> Example: Router(config-dial-peer)# destination-pattern 7777 | Specifies the directory number that the system uses to create music on hold. This command specifies either the prefix or the full E.164 telephone number to be used for a dial peer. |
| Step 3 | <code>port port</code> Example: Router(config-dial-peer)# port 1/1/0 | Associates the dial peer with the voice port that was specified in the “Setting Up the Voice Port on the Cisco SRST Gateway” section on page 26. |
| Step 4 | <code>exit</code> Example: Router(config-dial-peer)# exit | Exits dial-peer configuration mode. |

Establishing the MOH Feed

To establish the MOH feed, connect the music source, such as a CD player to autodial the directory number.

SUMMARY STEPS

1. **call-manager-fallback**
2. **max-dn** *max-directory-number*
3. **multicast moh** *multicast-address* **port** *port* [**route** *ip-address-list*]
4. **moh-live dn-number** *calling-number* **out-call** *outcall-number*
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 | max-dn <i>max-directory-number</i> Example: Router(config-cm-fallback)# max-dn 1 | Sets the maximum possible number of virtual voice ports that can be supported by a router. <ul style="list-style-type: none">• <i>max-directory-number</i>—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum possible number is platform-dependent. The default is 0. |

| Command or Action | Purpose |
|--|---|
| <p data-bbox="142 331 212 363">Step 3</p> <pre data-bbox="240 331 776 390">multicast moh <i>multicast-address</i> port <i>port</i> [route <i>ip-address-list</i>]</pre> <p data-bbox="240 436 342 468">Example:</p> <pre data-bbox="240 468 776 548">Router(config-cm-fallback)# multicast moh 10.10.255.250 port 2 route 10.10.255.251 10.10.255.252 10.10.255.253 10.10.255.254</pre> | <p data-bbox="862 331 1539 390">Enables multicast of MOH from a branch office flash MOH file to IP phones in the branch office.</p> <p data-bbox="862 411 1539 506">Note This command must be used to source live feed MOH to multicast Cisco CallManager mode. It is not required in strict SRST mode.</p> <ul data-bbox="878 527 1539 1146" style="list-style-type: none"> <li data-bbox="878 527 1539 915">• <i>multicast-address</i> and <i>port port</i>—Declares the IP address and port number of MOH packets that are to be multicast. The multicast IP address and port must match the IP address and the port number that Cisco CallManager is configured to use for multicast MOH. If you are using different codecs for MOH, these may not be the base IP address and port, but an incremented IP address or port number. See the “Configuring the MOH Audio Source to Enable Multicasting” section on page 9. If you have multiple audio sources configured on Cisco CallManager, ensure that you are using the audio sources’ correct IP address and port number. <li data-bbox="878 926 1539 1146">• <i>route ip-address-list</i>—(Optional) Declares the IP address or addresses from which the flash MOH packets can be transmitted. A maximum of four IP address entries are allowed. If a <i>route</i> keyword is not configured, the Cisco Unified SRST system uses the <i>ip source-address</i> command value configured for Cisco SRST. |
| <p data-bbox="142 1161 212 1192">Step 4</p> <pre data-bbox="240 1161 776 1220">moh-live dn-number <i>calling-number</i> out-call <i>outcall-number</i></pre> <p data-bbox="240 1266 342 1297">Example:</p> <pre data-bbox="240 1297 776 1356">Router(config-cm-fallback)# moh-live dn-number 3333 out-call 7777</pre> | <p data-bbox="862 1161 1539 1220">Specifies that this telephone number is to be used for an outgoing call that is to be the source for an MOH stream.</p> <ul data-bbox="878 1241 1539 1514" style="list-style-type: none"> <li data-bbox="878 1241 1539 1335">• <i>dn-number calling-number</i>—Sets the MOH telephone number. The <i>calling-number</i> is a sequence of digits that represent a telephone number. <li data-bbox="878 1346 1539 1514">• <i>out-call outcall-number</i>—Indicates that the router is calling out for a live feed that is to be used for MOH and specifies the number to be called. The <i>outcall-number</i> is a sequence of digits that represent a telephone number, typically of an E&M port. <p data-bbox="911 1524 1539 1640">The <i>outcall</i> keyword makes a connection to the local router voice port that was specified in the “Setting Up the Voice Port on the Cisco SRST Gateway” section on page 26.</p> |
| <p data-bbox="142 1654 212 1686">Step 5</p> <pre data-bbox="240 1654 302 1686">exit</pre> <p data-bbox="240 1732 342 1764">Example:</p> <pre data-bbox="240 1764 659 1797">Router(config-cm-fallback)# exit</pre> | <p data-bbox="862 1654 1539 1686">Exits call-manager-fallback configuration mode.</p> |

Verifying Cisco SRST MOH Live Feed

To verify MOH live feed, use the **debug ephone moh** command and the other commands described in the [“Verifying Basic Cisco SRST Multicast MOH Streaming”](#) section on page 20.

Configurations Examples for Cisco SRST Gateways

This section provides the following configuration example for Cisco SRST gateways:

- [MOH Routed to Two IP Addresses: Example, page 30](#)
- [MOH Live Feed: Example, page 30](#)

MOH Routed to Two IP Addresses: Example

The following example declares the Cisco CallManager multicast MOH IP address 239.1.1.1 and port number 16384 and streams music-on-hold.au audio file packets out the interfaces that are configured with the IP addresses 10.1.1.1 and 172.21.51.143.

```
ccm-manager music-on-hold
interface Loopback0
 ip address 10.1.1.1. 255.255.255.255

interface FastEthernet0/0
 ip address 172.21.51.143 255.255.255.192

call-manager-fallback
 ip source-address 172.21.51.143 port 2000
 max-ephones 1
 max-dn 1
 moh music-on-hold.au
 multicast moh 239.1.1.1 port 16384 route 172.21.51.143 10.1.1.1
```



Note

The multicast IP address and port must match the IP address and the port number that Cisco CallManager is configured to use for multicast MOH. If you are using different codecs for MOH, these may not be the base IP address and port, but an incremented IP address or port number. See the [“Configuring the MOH Audio Source to Enable Multicasting”](#) section on page 9. If you have multiple audio sources configured on Cisco CallManager, ensure that you are using the audio source’s correct IP address and port number.

MOH Live Feed: Example

The following example configures MOH from a live feed. Note that the dial peer references the E&M port that was set with the **voice-port** command and that the dial peer number (7777) matches the outcall number configured with the **out-call** keyword of the **moh-live** command.

```
.
.
.
```

```

voice-port 1/0/0
  input gain 3
  auto-cut-through
  operation 4-wire
  signal immediate
!
dial-peer voice 7777 pots
  destination-pattern 7777
  port 2/0/0
!
!
call-manager-fallback
  max-conferences 8
  max-dn 1
  moh-live dn-number 3333 out-call 7777
!
.
.
.

```

Additional References

The following sections provide references related to Cisco SRST multicast MOH.

Related Documents

| Related Topic | Document Title |
|---|---|
| Cisco SRST command reference | Cisco IOS Survivable Remote Site Telephony Version 3.2 Command Reference |
| Cisco IP phones | Cisco IP Phones and Services |
| Command reference and configuration information for voice and SRST | Cisco IOS Voice Command Reference, Release 12.3T Cisco IOS Debug Command Reference, Release 12.3T Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide Cisco SRST 3.2 Command Reference |
| Cisco CallManager user documentation, including the <i>System Administration and Features and Services Guides</i> | Cisco CallManager |
| Cisco CallManager music on hold | The “Music On Hold” chapter of the <i>Cisco CallManager Features and Services Guide</i> for your specific Cisco CallManager release. From the Cisco CallManager documentation directory, click the desired Cisco CallManager version and go to System Administration and Features and Services > Cisco CallManager Administration, System, and Features and Services Guides > Cisco CallManager Features and Services Guide > Music On Hold . |

Standards

| Standards | Title |
|--|-------|
| No new or modified standards are supported by this feature, and support for existing MIBs has not been modified by this feature. | — |

MIBs

| MIBs | MIBs Link |
|---|--|
| No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature. | To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs |

RFCs

| RFCs | Title |
|---|-------|
| No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature. | — |

Technical Assistance

| Description | Link |
|--|---|
| Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content. | http://www.cisco.com/public/support/tac/home.shtml |