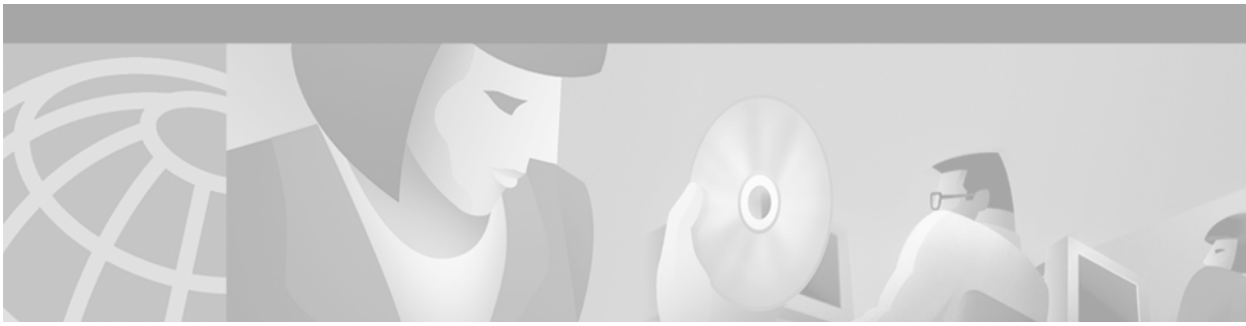


Voice Multicasting on Cisco 2600 Series and Cisco 3600 Series Routers

Feature Overview

The voice multicasting feature on Cisco 2600 and Cisco 3600 series routers uses Cisco voice over IP technology to create a permanently connected point-to-multipoint hoot-and-holler network over an IP connection. Hoot and holler is a broadcast audio network used extensively by the brokerage industry for market updates and trading. Similar networks are also used in publishing, transportation, power plants, and manufacturing.

You can connect voice multicasting telephones to network routers in any of the following ways:

- Connect a four-wire E&M telephone, which has no dial and is always off-hook, directly to an E&M voice interface card installed in a voice network module. Configure the E&M interface for four-wire trunk operation. For information about configuring E&M interfaces, see the *Cisco IOS Release 12.0 Voice, Video, and Home Applications Configuration Guide.*
- Connect a conventional telephone to a PBX that is connected to an E&M voice interface card.

Note Voice multicasting over FXS and FXO voice interface cards is not supported at this time.

- Connect a conventional telephone to a PBX that is connected through a T1 line to a multiflex trunk interface card installed in a high-density voice network module.

Note The voice multicasting feature supports only one T1 line per high-density voice network module.

Benefits

Hoot and holler and similar networks can gain significant benefits by running over an IP network, since any idle bandwidth can be reclaimed by data applications.

Related Documents

For information about installing voice network modules and voice interface cards in Cisco 2600 series and Cisco 3600 series routers, see these publications:

- *Cisco Network Module Hardware Installation Guide*
- *WAN Interface Card Hardware Installation Guide*

For information about configuring voice over IP features, see these publications:

- *Software Configuration Guide for Cisco 3600 Series and Cisco 2600 Series Routers*
- *Voice over IP Quick Start Guide*
- *Cisco IOS Release 12.0 Voice, Video, and Home Applications Configuration Guide*

For further information about IP multicasting, see this site:

- *IP Multicast Site* (http://www.cisco.com/ipmulticast)

Supported Platforms

Voice multicasting is supported on the Cisco 2600 series and Cisco 3600 series of modular routers.

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

No new or modified MIBs are supported by this feature.

For descriptions of supported MIBs and how to use MIBs, see the Cisco MIB web site on CCO at http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.

RFCs

No new or modified RFCs are supported by this feature.

Configuration Tasks

See the following sections for configuration tasks:

- Configuring Voice Ports (Required)
- Configuring Voice Ports in High-Density Voice Network Modules (Required)
- Configuring Dial Peers (Required)
- Configuring Ethernet (Required)
- Configuring Quality of Service (Optional)

Configuring Voice Ports (Required)

	Command	Purpose
Step 1	`Router(config)# ip multicast-routing`	Enable multicast routing.
Step 2	`Router(config)# voice class permanent` *tag1*	Define voice class for transmit-receive mode.
Step 3	`Router(config-class)# signal timing oos timeout disabled`	Disable signaling loss detection.
Step 4	`Router(config-class)# signal keepalive` *number*	Specify keepalive signaling packet interval.
Step 5	`Router(config-class)# voice class permanent` *tag2*	Define voice class for receive-only mode.
Step 6	`Router(config-class)# signal timing oos suppress-all` *seconds*	If the transmit out-of-service pattern (from the PBX to the network) matches for the time specified, the router stops sending packets to the network.
Step 7	`Router(config-class)# signal keepalive` *number*	Specify keepalive signaling packet interval.
Step 8	`Router(config)# interface` *virtual-interface*	Define a virtual interface for multicast fast switching. Routers joining the same session must have their virtual interfaces on different subnets. Otherwise packets are not switched to the IP network.
Step 9	`Router(config-if)# ip address` *address subnet-mask*	Assign the IP address and subnet mask for the virtual interface.
Step 10	`Router(config-if)# ip pim dense-mode`	Specify Protocol Independent Multicast (PIM) dense-mode.
Step 11	`Router(config)# voiceport router-`*slot/voice-slot/VIC-port*	Select the voice port to configure.
Step 12	`Router(config-voiceport)# voice-class permanent` *tag1*	Use voice class *tag1* for the port that is allowed to speak.
Step 13	`Router(config-voiceport)# vad`	Enable voice activity detection (VAD). This is the default setting and should not be changed.
Step 14	`Router(config-voiceport)# connection trunk` *phone-number*	Tie the voice port to a phone number.
Step 15	`Router(config-voiceport)# music-threshold` *threshold*	Set the music threshold to make VAD less sensitive.
Step 16	`Router(config-voiceport)# operation 4-wire`	Specify 4-wire operation.
Step 17	`Router(config-voiceport)# voiceport router-`*slot/voice-slot/VIC-port*	Select another voice port.
Step 18	`Router(config-voiceport)# voice-class permanent` *tag2*	Use voice class *tag2* for the receive-only port.
Step 19	`Router(config-voiceport)# vad`	Enable VAD.
Step 20	`Router(config-voiceport)# connection trunk` *phone-number*	Tie the voice port to the same phone number as in Step 14.
Step 21	`Router(config-voiceport)# music-threshold` *threshold*	Set the music threshold to make VAD less sensitive.
Step 22	`Router(config-voiceport)# operation 4-wire`	Specify 4-wire operation.

Configuring Voice Ports in High-Density Voice Network Modules (Required)

A multiflex trunk interface card in a high-density voice network module requires special voice-port configuration.

	Command	Purpose
Step 1	`Router(config)# voice-card` *number*	Select the card to configure.
Step 2	`Router(config-voicecard)# codec complexity high`	Codec complexity must be high. Voice multicasting does not support medium complexity, which is the default.
Step 3	`Router(config)# controller t1` *slot/port*	Select the T1 controller to configure.
Step 4	`Router(config-controller)# ds0-group` *ds0-group-number* `timeslots` *timeslot-list* `type e&m-immediate-start`	Map each DS0 group to a timeslot with the same number. This command is repeated for each group from 1 to 23.
Step 5	`Router(config)# voice-port` *slot/port*`:`*ds0-group-number*	Map each DS0 to voice port *slot/port:ds0-group-number*. This command is repeated for each group number from 1 to 23.
Step 6	`Router(config-voiceport)# connection trunk` *phone-number*	Tie the connection trunk to a phone number. This command is repeated for each DS0 group. All groups use the same phone number.

Configuring Dial Peers (Required)

	Command	Purpose
Step 1	`Router(config)# dial-peer voice` *tag* `voip`	Assign a tag to the VOIP dial peer.
Step 2	`Router(config-dial-peer)# destination-pattern` *phone-number*	The destination pattern for the VOIP dial peer must match the connection trunk string for the corresponding voice port.
Step 3	`Router(config-dial-peer)# session protocol multicast`	Enable multicasting. This step is mandatory for voice multicasting.
Step 4	`Router(config-dial-peer)# session target ipv4:`*address*`:`*port*	Assign the session target for voice multicasting dial peers. This is a multicast address in the range 224.0.1.0 to 239.255.255.255, and must be the same for all ports in a session. The audio RTP port is an even number in the range 16384 to 32767, and must also be the same for all ports in a session.
Step 5	`Router(config-dial-peer)# ip precedence` *number*	Specify the IP precedence.
Step 6	`Router(config-dial-peer)# codec {g711alaw | g711ulaw | g726r32 | g729ar8 | g729r8}`	Configure the codec. You must configure the same codec on all dial peers in a session. Only G.711, G.726, and G.729 codecs are supported. When the default codec, G.729, is used, it does not appear in the configuration.

Configuring Ethernet (Required)

	Command	Purpose
Step 1	`Router(config)# interface ethernet` *slot/port*	Configure the physical interface for transmitting multicast packets.
Step 2	`Router(config-if)# ip address` *address subnet-mask*	Assign the IP address and subnet mask for the interface.
Step 3	`Router(config-if)# ip pim sparse-dense-mode`	PIM should always be configured for sparse-dense-mode.
Step 4	`Router(config-if)# ip sap listen`	Listen to packets of Session Announcement Protocol.
Step 5	`Router(config-if)# ip igmp join-group` *address*	The address in this command must match the multicast address (session target) for the session.
Step 6	`Router(config-if)# no shutdown`	Enable the interface.

Configuring Quality of Service (Optional)

Voice traffic is much more sensitive to timing variations than data traffic. For good voice performance over a WAN, you might need to configure your data network so voice packets are not lost or delayed. This section shows how to improve quality of service (QoS) for voice multicasting over a Frame Relay serial connection.

	Command	Purpose
Step 1	`Router(config)# interface serial slot/port`	Specify the interface to configure.
Step 2	`Router(config-if)# encapsulation frame-relay`	Configure Frame Relay encapsulation.
Step 3	`Router(config-if)# frame-relay traffic-shaping`	Configure Frame Relay traffic shaping.
Step 4	`Router(config-if)# no frame-relay broadcast-queue`	Disable the broadcast queue.
Step 5	`Router(config-if)# interface serial slot/port.subinterface point-to-point`	Specify the subinterface to configure.
Step 6	`Router(config-if)# ip address subnet-mask`	Assign an IP address and subnet mask.
Step 7	`Router(config-if)# ip pim sparse-dense-mode`	Configure PIM sparse-dense mode.
Step 8	`Router(config-if)# frame-relay class name`	Specify the Frame Relay map class to associate with this subinterface.
Step 9	`Router(config-if)# frame-relay interface-dlci number`	Assign a DLCI to the interface.
Step 10	`Router(config-if)# frame-relay ip rtp header-compression`	Enable IP RTP header compression.
Step 11	`Router(config-if)# map-class frame-relay name`	Create the map class to be associated with the subinterface.
Step 12	`Router(config-map-class)# frame-relay cir bps`	Specify the committed information rate (CIR).
Step 13	`Router(config-map-class)# frame-relay bc bits`	Specify the committed burst size.
Step 14	`Router(config-map-class)# frame-relay mincir bps`	Specify the minimum acceptable CIR>
Step 15	`Router(config-map-class)# no frame-relay adaptive-shaping`	Disable adaptive traffic shaping.
Step 16	`Router(config-map-class)# frame-relay fair-queue`	Enable weighted fair queueing.
Step 17	`Router(config-map-class)# frame-relay fragment fragment_size`	Enable fragmentation of Frame Relay frames.
Step 18	`Router(config-map-class)# frame-relay ip rtp priority audio-port number-of-ports bandwidth`	The first number is the audio port. The second number is the number of consecutive audio ports to which the IP RTP priority queuing applies. The third number is the bandwidth, which should equal the bandwidth needed for each call multiplied by the number of calls.

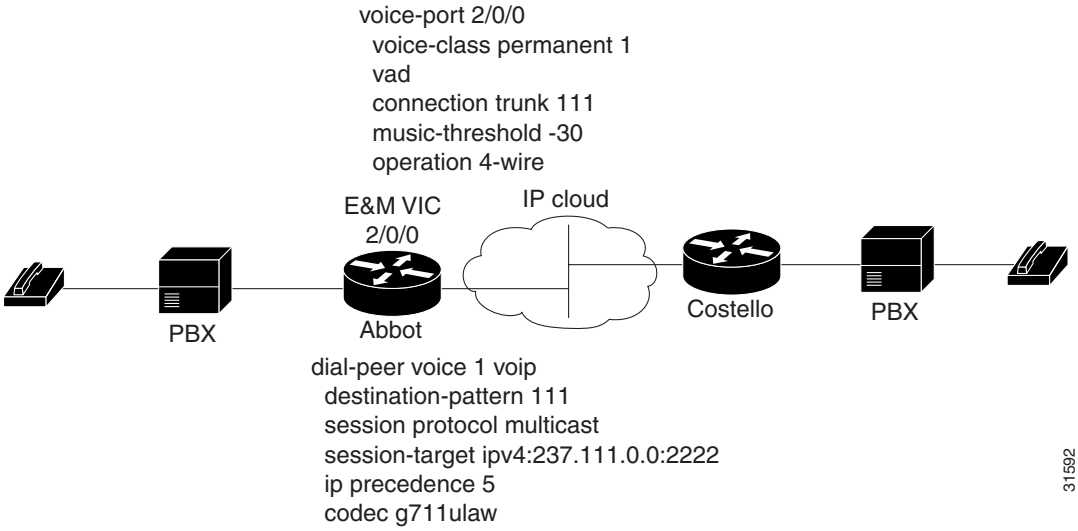
Configuration Examples

This section provides a series of configuration examples that help you to become familiar with voice multicasting. These examples also tell you how to ensure that each configuration is working properly before proceeding to the next step.

Voice Multicasting over an Ethernet LAN (One Session)

Figure 1 shows the simplest configuration. Two routers are connected to each other over an Ethernet LAN. One E&M phone is connected to each router.

Figure 1 Voice Multicasting over a LAN (One Session)



Voice Port Configuration

In router Abbott, the phone is connected to voice port 2/0/0, using the *router-slot/voice-slot/VIC-port* numbering convention. This voice port is configured as follows:

```
hostname abbott
!Enable multicast routing.
!
ip multicast-routing
!
!Define voice class for transmit-receive mode with tag 1.
!Disable signaling loss detection.
!Send keepalive packet every 65 seconds.
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
!Define voice class for receive-only mode with tag 2.
!
voice class permanent 2
signal timing oos suppress-all 1
signal keepalive 65535
!
!Define virtual interface for multicast fast switching.
!Routers joining the same session should have the virtual interfaces
!on different subnets. Otherwise packets will not be switched to the IP network.
!
interface vif1
ip address 1.1.1.1 255.255.255.0
ip pim dense-mode
!
!Configure voice ports.
!Use voice class tag 1 for port that is allowed to speak.
!Use voice class tag 2 for listen-only port.
!Set music threshold to make VAD less sensitive. Only noise above
!-30 dB is considered voice.
!Tie voice port to phone number 111, joining multicast session 237.111.0.0:22222.
!Joining session 111.
!
voice-port 2/0/0
voice-class permanent 1
vad
connection trunk 111
music-threshold -30
operation 4-wire
!
!Joining session 111 in receive-only mode.
!
voice-port 2/0/1
voice-class permanent 2
vad
connection trunk 111
music-threshold -30
operation 4-wire
!
```

The connection-trunk connection type is a point-to-point connection, similar to a tie-line on a PBX network. All voice traffic, including signaling, placed at one end is immediately transferred to the other.

The voice port must be configured for 4-wire operation.

High-Density Voice Modules

A multiflex trunk interface card in a high-density voice network module requires special voice-port configuration. First select the card to configure:

```
voice-card 6
 codec complexity high
!
```

Note Codec complexity must be high. Voice multicasting does not support medium complexity, which is the default.

The following commands define the T1 channel and signaling method, and map each DS0 to voice port *slot/port:ds0-group*:

```
controller T1 6/0
 ds0-group 1 timeslots 1 type e&m-immediate-start
 ds0-group 2 timeslots 2 type e&m-immediate-start
 ds0-group 3 timeslots 3 type e&m-immediate-start
 …
 ds0-group 22 timeslots 22 type e&m-immediate-start
 ds0-group 23 timeslots 23 type e&m-immediate-start
```

These commands configure the voice ports on the multiflex trunk interface card:

```
!
voice-port 6/0:1
 connection trunk 999
!
voice-port 6/0:2
 connection trunk 999
!
voice-port 6/0:3
 connection trunk 999
 …
voice-port 6/0:22
 connection trunk 999
!
voice-port 6/0:23
 connection trunk 999
```

Dial Peer Configuration

Cisco IOS software uses objects called dial peers to tie together telephone numbers, voice ports, and other call parameters. Configuring dial peers is similar to configuring static IP routes—you are telling the router what path to follow to route the call.

Dial peers are identified by numbers, but to avoid confusing these numbers with telephone numbers, they are usually referred to as tags. Dial peer tags are integers that can range from 1 to 2^{31} -1 (2147483647). Dial peers on the same router must have unique tags, but you can reuse the tags on other routers.

The following commands configure a dial peer with tag 1 for this voice port:

```
!Configure dial peer.
!Conference 1.
!Phone number 111.
!Multicast address 237.111.0.0, udp port 22222.
dial-peer voice 1 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.0:22222
ip precedence 5
 codec g711ulaw
!
```

Tips

- Note that the destination pattern 111 for the VOIP dial peer matches the connection trunk string for the corresponding voice port.
- The **session protocol multicast** command is essential for voice multicasting.
- The session target for voice multicasting dial peers is a multicast address in the range 224.0.1.0 to 239.255.255.255. This session target must be the same for all ports in a session. The audio RTP port is an even number in the range 16384 to 32767, and must also be the same for all ports in a session.
- Note the following restrictions on codecs:
  - You must configure the same codec on all dial peers in a session.
  - Only G.711, G.726, and G.729 codecs are supported.
  - When the default codec, G.729, is used, it does not appear in the configuration.
- Voice activity detection (VAD) is enabled by default. This setting should not be changed.

Ethernet Configuration

Configure the router's Ethernet interface as follows:

```
!Configure physical interface for transmitting multicast packets.
!
interface ethernet 0/0
ip address 1.5.13.13 255.255.255.0
ip pim sparse-dense-mode
ip sap listen
ip igmp join-group 237.111.0.0
no shutdown
!
```

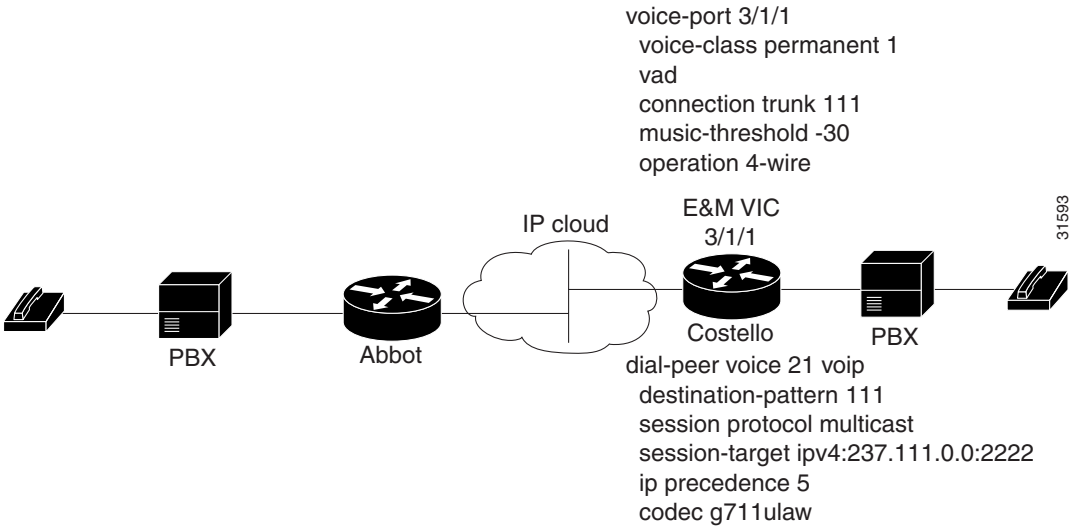
Tips

- PIM should always be configured for sparse-dense-mode.
- The address in the **ip igmp join-group** command must match the multicast address for the session.

Configuring the Second Router

- In router Costello, the E&M phone is connected to voice port 3/1/1. Router Costello uses the same configuration as Abbott, except for the following differences:
- The virtual interface must be on a different subnet from the first router.
- The IP address in the Ethernet configuration must be different.
- The voice port and slot should match the router's hardware configuration.

Figure 2 Voice Multicasting over a LAN (Second Router)



Tips

- The multicast session for this port, shown in the **session target** and **ip igmp join-group** commands, matches the multicast session configured on the first router.
- The codec configured for this dial peer matches the codec for the dial peer on the first router.
- Both routers are configured to use the same connection trunk and destination pattern.

Checking the Configuration

If you configured your routers following these examples, you should now be able to talk over the telephones. You can also use the **show dial-peer voice** command on each router to verify that the data you configured is correct.

To verify that an audio path has been established, use the **show call active voice** command. This command displays all active voice calls traveling through the router.

Voice Multicasting over a WAN

The configuration for voice multicasting sessions over IP on a Frame Relay, ATM, or other WAN is exactly the same as for the Ethernet LAN in the last example. Configure the WAN interface on each router with the **ip address**, **ip igmp join-group**, and **ip pim sparse-dense-mode** commands as shown in that example.

Quality of Service

Voice traffic is much more sensitive to timing variations than data traffic. For good voice performance, you might need to configure your data network so voice packets are not lost or delayed. The following example shows one way to improve quality of service (QoS) for voice multicasting over a Frame Relay connection:

```
!Configure physical interface for transmitting multicast packets.
!Listen to packets of Session Announcement Protocol.
!This example uses a subinterface
!
interface serial0/0
 encapsulation frame-relay
 frame-relay traffic-shaping
 no frame-relay broadcast-queue
!
interface serial0/0.1 point-to-point
 ip address 5.5.5.5 255.255.255.0
 ip pim sparse-dense-mode
 frame-relay class hootie
 frame-relay interface-dlci 100
 frame-relay ip rtp header-compression
!
!Frame relay class commands.
!
map-class frame-relay hootie
 frame-relay cir 64000
 frame-relay bc 2000
 frame-relay mincir 64000
 no frame-relay adaptive-shaping
 frame-relay fair-queue
 frame-relay fragment 80
 frame-relay ip rtp priority 16384 16383 64
```

Note In the **frame-relay ip rtp** priority command, the first number is the audio port. The second number is the number of consecutive audio ports to which the IP RTP priority queuing applies. The third number is the bandwidth, which should equal the bandwidth needed for each call multiplied by the number of calls.

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command reference publications.

session protocol multicast

To set the session protocol as multicast, use the **session protocol multicast** dial-peer configuration command.

session protocol multicast

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced.