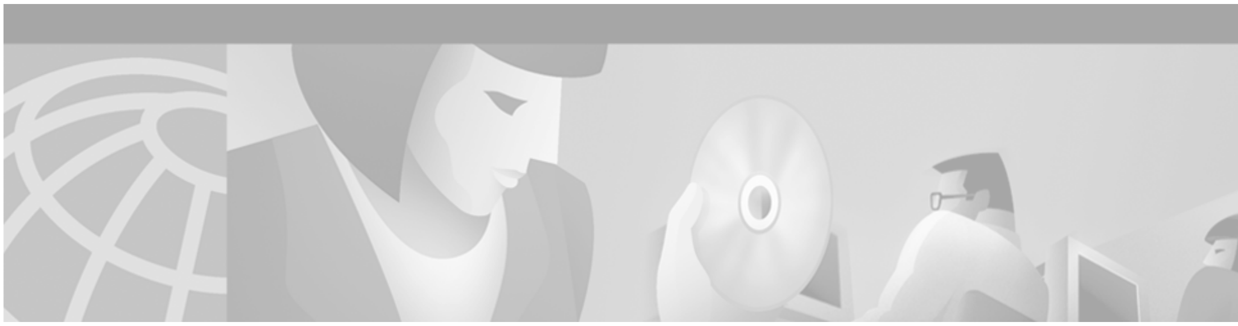


Cisco Hoot and Holler over IP

The voice multicasting feature on Cisco 2600 and Cisco 3600 series routers uses Cisco Voice-over-IP (VoIP) technology to create a permanently connected point-to-multipoint hoot and holler network over an IP connection.

Four-wire E&M, E1/T1, FXO, and FXS configurations provide continuous VoIP connections across a packet network using the connection-trunk mechanism. By using the inherent point-to-multipoint connectivity of IP multicast (IPmc), the routers can take several inbound voice streams from the traditional hoot devices and forward the packetized voice over the IP network to all parties within a defined hoot and holler group.

This feature module describes the Cisco Hoot and Holler over IP feature and contains the following sections:

Feature Overview

Hoot and holler networks provide "always on" multiuser conferences without requiring that users dial into a conference. These networks came into being more than 40 years ago when local concentrations of small specialized businesses with common, time-critical informational interests—junkyards, for example—began to install their own phone wires, speakers (called “squawk boxes”), and microphones between their businesses to ask each other about parts customers needed. These networks functioned as crude, do-it-yourself, business-to-business intercom systems.

Hoot and holler broadcast audio network systems have since evolved into the specialized leased-line networks used by financial and brokerage firms to trade stocks and currency futures and the accompanying time-critical information such as market updates and morning reports.

Users of various forms of hoot and holler networks now include brokerages, news agencies, publishers, weather bureaus, transportation providers, power plant operators, manufacturers, collectibles dealers, talent agencies, and nationwide salvage yard organizations.

Hoot and holler is used in these various industries as a way to provide a one-to-many or many-to-many conferencing service for voice communications. In the past, hoot and holler was deployed using point-to-point telco circuits and a hoot and holler bridging and mixing functionality that was provided either by the customer or as a service of the Public Switched Telephone Network (PSTN) carrier.

A common use of hoot and holler is a broadcast audio network that is used throughout the brokerage industry to communicate morning reports as well as to advise the trading community within a brokerage firm on market movements, trade executions, and so on. All users can talk simultaneously with each other, if desired.

But more commonly, a broker in a field office will “shout” an order to the trading floor. The shout ensures that the trading floor can hear the order and a floor trader can confirm the transaction. A typical brokerage firm has several of these networks for equity, retail, and bonds with network size and degree of interactivity varying depending on the application.

Within the financial community there are two general uses for hoot and holler networks:

- Market updates—Market update (morning report) hoot networks tend to be active for an hour in the morning and inactive for the rest of the day.
- Trading—Trading hoot networks tend to be far more used throughout the trading day.

Both of these applications can reap significant advantages by running over an IP network because any idle bandwidth can be reclaimed by data applications.

Today most hoot and holler customers pay for separate leased line charges from a common carrier to transport their hoot and holler to remote branch offices. This recurring charge is usually significant—some larger firms spend more than $2-3 million per year just to distribute hoot and holler feeds.

Current Hoot and Holler Implementations

Traditional hoot and holler networks (Figure 1) are analog, multipoint, 4-wire, audio-conference networks that are always up. When a user wants to communicate, the user pushes a button and speaks either through a microphone, a hoot phone, a turret, or a squawk box.

Figure 1 Traditional Hoot and Holler Network

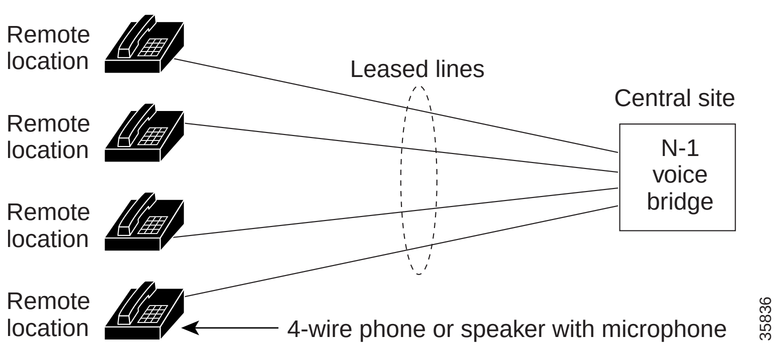


Figure 1 illustrates a traditional hoot and holler network. Each remote location is connected to a central bridge using leased lines. Four-wire connections and N-1 bridges are used to avoid echo problems.

Hoot and holler networks are typically spread over four to eight sites, although financial retail networks may have hundreds of sites interconnected. Within a site, bridging (mixing voice signals) is done locally with a standard analog or digital bridge that may be part of a trading turret system. Between sites, there are two prevalent methods for providing transport:

- Point-to-point leased lines with customer-provided audio bridging at a central site, and
- Carrier-provided audio bridging.

When customers provide their own bridging services with point-to-point leased lines, branch offices in a metropolitan area commonly have 25 to 50 lines or more.

The second method, carrier-provided audio bridging, is prevalent within the United States but rare for overseas transport. In this scenario, the audio bridges are located at the carrier's central office and the 4-wire lines are terminated at the client's site on a local audio-bridge equipped with 4-wire plug-ins, which then feed to local PA system speakers. Customer-provided hoot bridging services can now be replaced with Cisco Hoot and Holler over IP solutions.

An Overview of Cisco Hoot and Holler over IP

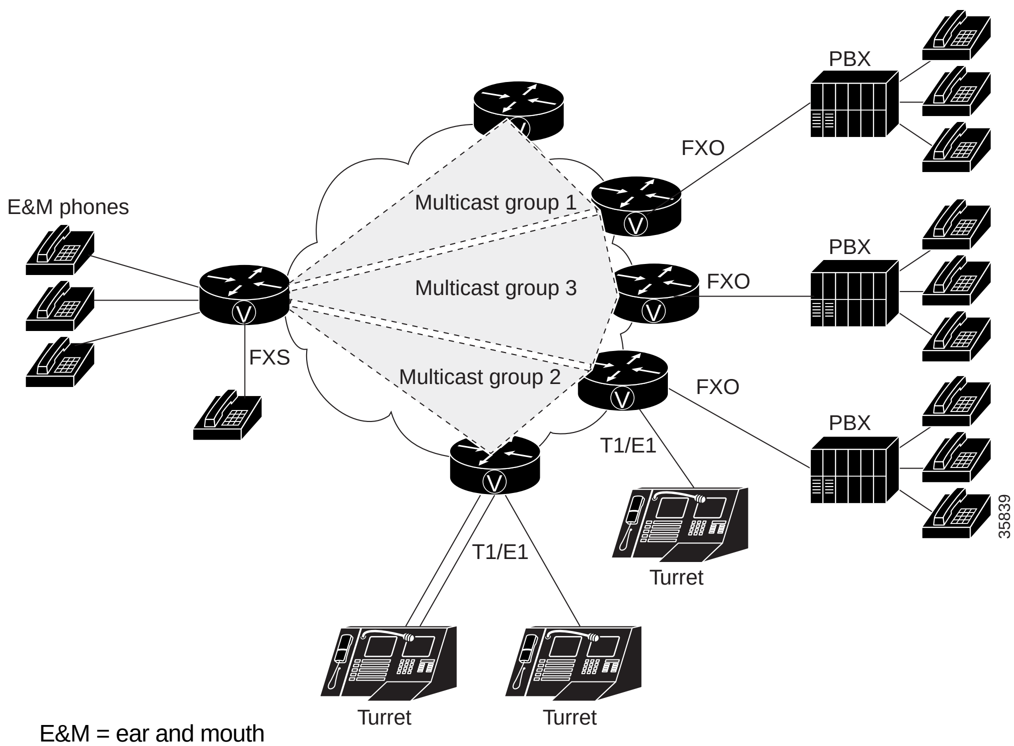
Cisco's VoIP technology, which was initially focused on traditional PBX toll-bypass applications, can be used to combine hoot and holler networks with data networks. While some customers may have done some level of hoot and data integration in the late 1980s with time-division multiplexing (TDM), this form of integration does not allow for dynamic sharing of bandwidth that is characteristic of VoIP. This dynamic sharing of bandwidth is even more compelling with hoot and holler than with a toll-bypass application, because some hoot circuits may be active for an hour or two for morning reports but might be dead for the rest of the day—the idle bandwidth can be used by the data applications during these long periods of inactivity.

Beginning with the Cisco IOS Release 12.1(2)XH, Cisco Hoot and Holler over IP can be implemented using Cisco's VoIP technology. This solution leverages Cisco's IOS expertise in VoIP, quality of service (QoS), and IP multicasting (IPmc) and is initially available on Cisco 2600 and 3600 series multiservice routers.

Figure 2 shows a diagram of the Cisco Hoot and Holler over IP solution connecting legacy hoot equipment over an IP network.

Note The "V" on the Cisco router icons signifies that some of the hoot and holler bridging function is being done by the router's digital signal processors (DSPs).

Figure 2 Hoot and Holler over IP using Cisco 2600 and Cisco 3600 Series Routers



Four-wire E&M, E1/T1, FXO, and FXS configurations provide continuous VoIP connections across a packet network. By using the inherent point-to-multipoint characteristic of IPmc, the routers can take several inbound voice streams from the traditional hoot devices, and forward the packetized voice over the IP network to all parties within a defined hoot and holler group.

Voice Multicasting

The voice multicasting feature on Cisco 2600 and Cisco 3600 series routers uses Cisco Voice over IP (VoIP) technology to create a point-to-multipoint hoot and holler network over an IP connection.

Voice multicasting telephones can be connected to routers in the following ways:

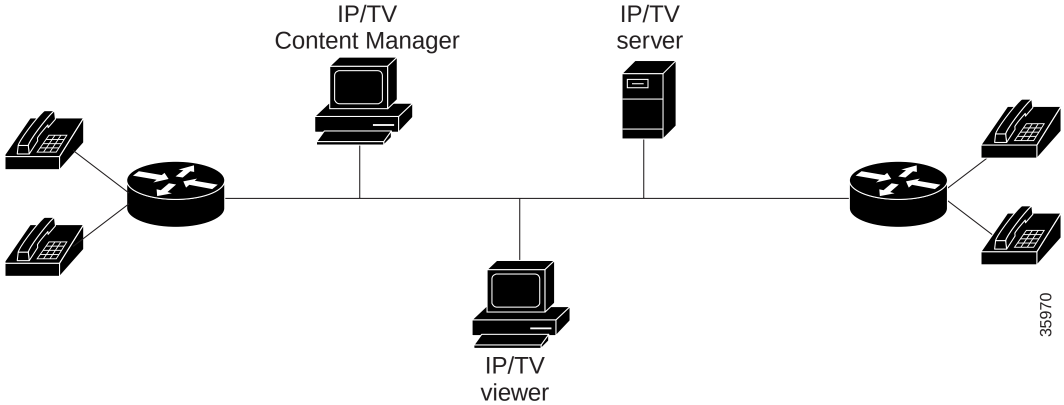
- Connect a 4-wire E&M telephone, which has no dial and is always off-hook, directly to an E&M voice interface card that is installed in a voice network module. Configure the E&M interface for four-wire trunk operation. For information about configuring E&M interfaces, see the *Cisco IOS Multiservice Applications Configuration Guide* , Release 12.1.
- Connect a conventional telephone to a PBX that is connected to an E&M voice interface card.
- Connect a conventional telephone to an FXS voice interface card that is installed in a voice network module.
- Connect a conventional telephone to a PBX that is connected through a E1/T1 line to a multiflex trunk interface card that is installed in a high-density voice network module.

Note The voice multicasting feature supports only one E1/T1 line per high-density voice network module.

IP/TV Access

The Cisco Hoot and Holler over IP feature enables you to access ongoing IP/TV multicasts for listening to voice content of the IP/TV session. For complete information on IP/TV, see the *IP/TV Content Manager Installation and User Guide* .

Figure 3 Cisco Hoot and Holler over IP Access to IP/TV Multicast



For the Cisco Hoot and Holler over IP and IP/TV interaction to work correctly:

- Ensure that you have a properly connected and configured network for Voice over IP with the Cisco Hoot and Holler over IP feature enabled, using the **session protocol multicast** command.
- Ensure that the server configured with the IP/TV Content Manager is in the same Ethernet network as the Cisco Hoot and Holler over IP functionality.
- Ensure that the Cisco Hoot and Holler over IP multicast details are registered with the IP/TV Content Manager.

Note IP/TV support for Cisco Hoot and Holler over IP uses only G.711 u-law (mu-law) encoding.

IP/TV supports one audio stream for Cisco Hoot and Holler over IP.

IP/TV does not support arbitration and mixing.

Content Manager

On the configuration screen (**Administration Tool>Scheduled Programs>New Program>Configuration**), provide the following details:

- Multicast address
- RTP port—defined by the dial-peer in the router
- IP/TV server—IP address or name
- From the **Settings>Content Manager** option:
  - Click **Add New**.
  - Enter the IP/TV server name.

- The port number must be 80, because it is HTTP.
- Click **OK** and exit.

Note In the Content Manager, be sure to specify the multicast IP address and RTP port for the Cisco Hoot and Holler over IP session.

Interactive Voice Response (IVR)

The Cisco Hoot and Holler over IP feature can support Interactive Voice Response (IVR) as a means of authentication, authorization, and accounting (AAA) control. See the section “Configuring Interactive Voice Response (IVR) ” in the *Multiservice Applications Configuration Guide* and the command descriptions in the *Multiservice Applications Command Reference* for more information.

Migration Strategy

To aid troubleshooting and allow for regionalized hoot and holler conferences, most hoot and holler networks today are structured by interconnecting multiple, regional hoot networks with a centralized bridge. The regional hoot networks are built using either carrier-based multidrop circuits or point-to-point circuits bridged by the customer. All of these circuits are connected through patch panels that allow for these regional bridges to be connected for a larger corporate-wide conference call. This is typically done for the "morning call" that is broadcast to all locations, advising of market movements, recommendations, and commentary. Later in the day, the patch panel may be reconfigured to allow for local or regional conference bridges. This allows for multiple conference calls for various purposes, without provisioning multiple circuits. By segmenting the network into regions, troubleshooting is also easier because any audio disturbance, feedback, or level problems can be isolated to a smaller subset of remote offices for more specific troubleshooting.

The highly segmented nature of existing hoot and holler networks can be leveraged in the migration from legacy hoot technology to Cisco Hoot and Holler over IP. A small segment of the hoot network can be converted to Cisco Hoot and Holler over IP while preserving the operational procedures at the main office.

Note that the migration to Cisco Hoot and Holler over IP does not require replacing end-user equipment or central bridging equipment. The main impetus for this first phase of migration is to eliminate the recurring expense of carrier multidrop circuits or dedicated leased lines. By minimizing changes presented to the end user while realizing an attractive payback period on the capital costs, migration success is maximized.

As the entire hoot network converges with the data network, additional functionality can be introduced. Since the hoot and holler connections are now carried in standard multicast RTP packets, hoot channels can now be received by a soft client such as IP/TV, which can receive an IP multicast RTP stream. An alternate migration strategy is to use Cisco Hoot and Holler over IP technology initially as a backup for the existing hoot circuits within a region with a phased plan of cutting over to Cisco Hoot and Holler over IP as the primary transport while keeping the existing circuits as a backup for a predefined burn-in period.

Technical Details of the Cisco Hoot and Holler over IP Solution

This section describes how Cisco Hoot and Holler over IP works from a technical perspective. It covers design considerations in terms of IOS configurations and DSP mixing functionality as well as bandwidth planning and QoS, with the following assumptions:

1. That you have some level of Cisco IOS experience.
2. That you have some experience configuring QoS features with Cisco IOS. If not, please refer to the IOS documentation on CCO at:

   http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121cgcr/ip_c/index.htm
3. That you have some experience configuring VoIP with Cisco IOS. If not, please refer to the IOS documentation on CCO at:

   http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121cgcr/multi_c/mcprt1/mcdvoip.htm
4. That you have some experience configuring IP multicasting with Cisco IOS. If not, please refer to the documentation on CCO at:

   http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121cgcr/ip_c/ipcprt3/1cdmulti.htm
5. That you have a working IP network, with IP multicasting configured using the Cisco 2600 and Cisco 3600 series routers. If not, please refer to the documentation on CCO at:

   http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121cgcr/index.htm

   http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis2600/index.htm

   http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3600/index.htm
6. That you are familiar with Cisco IP/TV. If not, please refer to the documentation on CCO at:

   http://www.cisco.com/univercd/cc/td/doc/product/software/iptv30/
7. That you understand basic hoot and holler concepts and equipment.

IP Multicast and DSP Arbitration and Mixing

When deploying Cisco Hoot and Holler over IP, first consider how the voice streams are going to be mixing and distributed to other locations. This is done using a combination of two technologies:

- IP multicast (IPmc)
- DSP arbitration and mixing

Since hoot and holler is generally used to allow many people to simultaneously talk and listen to other people within a hoot group, by definition it requires that the same speech be delivered to multiple parties at the same time. In an IP network, this functionality uses IP multicasting (IPmc). IPmc allows a source to send a single packet into the IP network and have it duplicated and sent to many listeners by the other routers within the network. This technique is beneficial in that it does not require the source to know how many listeners there are, as well as not requiring additional processing burden on the source by having to send a copy of each packet to all listeners. IPmc also allows for listeners to dynamically join IPmc groups, which eliminates the administrative burden of adding new users every time a new IPmc session is initiated.

Now that we have established that an IP network can forward packets in a way similar to existing hoot and holler networks, we also must examine how the individual router/gateways can handle mixing and arbitrating the various voice streams that could initiate or terminate on its voice ports. This functionality is handled by the onboard DSPs on each voice-card (NM-1V, NM-2V or NM-HDV). Arbitration involves identifying the various sources of the voice stream, and mixing involves taking some of those voice streams and combining them into a single-sourced voice stream. Cisco Hoot and Holler over IP can handle many inbound voice streams, *but it only arbitrates and mixes three streams to be heard within the Hoot group*. This value works fine in most applications, because beyond three streams two things happen in normal human conversation:

1. People are not able to distinguish the content of more than three streams.
2. People normally stop speaking if they hear others talking ahead of them.

Note The mixing functionality does not do a summation of the voice streams.

As previously mentioned, the DSPs in Cisco Hoot and Holler over IP do mixing for up to three streams. This fact is important when network administrators consider how much bandwidth they should plan for in their Cisco Hoot and Holler over IP network. This is especially crucial when planning WAN bandwidth, which is often much more expensive and much less available than LAN bandwidth.

The advantage to this functionality is that a network administrator never has to be concerned about provisioning voice bandwidth for more than three times per call bandwidth for each WAN site, which helps to simplify overall network planning.

Bandwidth Planning

Four main factors must be considered with regard to bandwidth planning for Cisco Hoot and Holler over IP:

1. Codecs used for VoIP (G.711, G.726, G.729 and G.729a are currently supported).
2. Bandwidth management techniques.
3. The number of voice streams to be mixed.
4. The amount of guaranteed bandwidth available on the IP network. This includes both LAN and WAN bandwidth, and should take into consideration things such as Frame Relay CIR.

Codecs

By default, Cisco IOS sends all VoIP traffic (media, using RTP) at a rate of 50 packets per second. The packets include not only the voice sample, but also an IP, UDP, and RTP header. The IP/UDP/RTP header adds an additional 40 bytes to each packet. The amount of bandwidth each VoIP call consumes depends on the codec selected. The resulting bandwidths can be:

- G.729 or G.729a = 3000bytes * 8 bits = 24Kb/call
- G.726 = 6000bytes * 8 bits = 48Kb/call
- G.711 = 10000bytes * 8 bits = 80Kb/call

In addition to these calculations, network administrators must consider the Layer 2 headers (Frame Relay, PPP, Ethernet, and so on.) and add the appropriate number of bytes to each packet.

The following table, Table 1, assumes a payload size (bytes) of 20 ms samples per packet with 50 packets per second.

The value of *n* is equal to the number of voice streams in a session.

The uncompressed bandwidth includes IP/UDP/RTP headers (40 bytes) in the bandwidth calculation. Compressed RTP (cRTP) reduces the IP/UDP/RTP headers to between 2 to 4 bytes per packet. The calculation of compressed bandwidth below uses 4 bytes for a compressed IP/UDP/RTP header per packet.

Maximum RTCP bandwidth is five percent of the total RTP traffic in a hoot and holler session. Since the Cisco Hoot and Holler over IP application supports mixing of a maximum of three voice streams, the RTCP bandwidth is limited to five percent of three-voice-stream traffic.

In addition to the above, Layer 2 headers (Frame Relay, PPP, Ethernet, and so on) should be considered and added to the bandwidth calculation.

Table 1 Bandwidth Consideration Table

Codec	Payload Size (byte)	Bandwidth/ Voice Stream (Kbps)		RTCP Bandwidth per Cisco Hoot and Holler over IP Session (Kbps)	Example—One Voice Stream in a Session (Bandwidth in Kbps)	
		Uncompressed	Compressed		=(1)*n+(3)	=(2)*n+(3)
g.729	20	24	9.6	3.6	27.6	13.2
g.726	80	48	33.6	7.2	55.2	40.8
g.711	160	80	65.6	12.0	92.0	77.6

cRTP, Variable-Payload Sizes and VAD

Some network administrators may consider this amount of bandwidth per call unacceptable or outside the limits for which they can provide bandwidth, especially in the WAN. There are several options that network administrators have for modifying the bandwidth consumed per call:

1. RTP header compression (cRTP)
2. Adjustable byte-size of the voice payload
3. Voice activity detection (VAD)

IP/UDP/RTP headers add an additional 40 bytes to each packet, but each packet header is basically unchanged throughout the call. cRTP can be enabled for the VoIP calls, which reduces the IP/UDP/RTP headers to between 2 to 4 bytes per packet.

More detailed documentation on cRTP can be found on CCO at:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121cgcr/qos_c/qcprt6/qcdcrtp.htm

In addition to reducing the IP/UDP/RTP headers per packet, the network administrator also has the option of controlling how much voice payload is included in each packet. This is done using the **bytes** keyword and argument in a VoIP dial-peer. The following example shows a dial-peer configuration:

```
dial-peer voice 1 voip
 destination-pattern 4085551234
 codec g729r8 bytes 40
 session protocol multicast
 session target ipv4:239.10.108.252:20102
```

As the number of bytes per packet is modified, so too is the number of packets per second that are sent.

Voice activity detection (VAD) enables the DSPs to dynamically sense when there are pauses in a conversation. When these pauses occur, no VoIP packets are sent into the network. This significantly reduces the amount of bandwidth used per VoIP call, sometimes as much as 40 to 50 percent. When voice is present, then VoIP packets are again sent. When using Cisco Hoot and Holler over IP, VAD must be enabled to reduce the amount of processing of idle packets by the DSPs. In basic VoIP, VAD can be enabled or disabled, but since the DSPs also have to do arbitration and mixing, VAD must be disabled to reduce the DSPs processing load. In addition to enabling VAD (which is on by default), network administrators should modify the VAD parameters that sense background noise so that idle noise does not consume bandwidth.

This can be configured as in the following E&M port example:

```
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
voice-port 1/0/0
 voice-class permanent 1
 connection trunk 111
 music-threshold -30
 operation 4-wire
```

The configuration above is used for a voice-port that is in send/receive mode, and only noise above -30Db is considered voice.

Virtual Interface (Vif)

In all Cisco Hoot and Holler over IP implementations, the routers are configured with an "interface vif1." This is a virtual interface that is similar to a loopback interface—a logical IP interface that is always up when the router is active. In addition, it must be configured so the Cisco Hoot and Holler over IP packets that are locally mixed on the DSPs can be fast-switched along with the other data packets. This interface must reside on its own unique subnet, and that subnet should be included in the routing protocol updates (RIP, OSPF, and so on).

Connection Trunk

Cisco Hoot and Holler over IP provides an "always-on" communications bridge—end users do not need to dial any phone numbers to reach the other members of a hoot group. To simulate this functionality, Cisco IOS provides a feature called "connection trunk." Connection trunk provides a permanent voice call, without requiring any input from the end user, because all of the digits are internally dialed by the router/gateway.

With traditional VoIP usages of connection trunk, the call is mapped to a remote router/gateway, and all the H.323 signaling is handled dynamically when the trunk is established. With Cisco Hoot and Holler over IP, the connection trunk is established to the IP address of the IP multicast (IPmc) group that maps to the hoot group.

In addition, all negotiation of UDP ports for the audio stream is manually configured. The following example shows an E&M voice port connection trunk set up for Cisco Hoot and Holler over IP:

```
voice-port 1/0/0
 connection trunk 111
 music-threshold -30
 operation 4-wire
!
dial-peer voice 1 voip
 destination-pattern 111
 voice-class permanent 1
 session protocol multicast
 session target ipv4:237.111.0.0:22222
 ip precedence 5
```

In this example, the digits in the **connection trunk** *111* string match the destination pattern of the VoIP dial peer. Also, the session protocol is set to multicast and the session target is pointing to the IPmc group number with the UDP port (22222) predefined.

Benefits

Cisco Hoot and Holler over IP provides the following benefits:

- Eliminates yearly reoccurring circuit-switched telecom charges (toll-bypass).
- Eliminates the need for leased-lines and the accompanying charges.
- Reduces the need for Hoot and Holler bridges.
- Improves Hoot and Holler network manageability.
- Reduces the time to troubleshoot a problem from hours to minutes.
- Reduces the time to provision bandwidth from days to a few hours.
- Increases productivity through future applications (such as IP/TV and turret support).
- Ability to integrate voice, video, and data signaling capabilities.

Related Documents

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

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

For information about configuring Voice over IP features, see the following publications:

- *Software Configuration Guide for Cisco 3600 Series and Cisco 2600 Series Routers*
- *Voice over IP Quick Start Guide*
- *Cisco IOS Multiservice Applications Configuration Guide* , Release 12.1

For further information about IP multicasting, go to this site:

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

For further information about IP/TV, see the following publication:

- *IP/TV Content Manager User Guide*

For further information about interactive voice response (IVR), see the following document:

- *Configuring Interactive Voice Response for Cisco Access Platforms*

Restrictions

- Cisco Hoot and Holler over IP supports the mixing of only three voice streams.
- IP/TV does not support the mixing of audio streams.
- IP/TV supports only G.711 u-law (mu-law).
- Voice Interface Card Basic Rate Interface (VIC-BRI) is not supported.

Supported Platforms

Router Platforms:

- Cisco 2600
- Cisco 3600 series

Network Modules:

- NM-HDV
- NM-1V
- NM-2V

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.

To obtain lists of MIBs supported by platform and Cisco IOS release and to download MIB modules, go to the Cisco MIB web site on Cisco Connection Online (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 configuring Cisco Hoot and Holler over IP:

- Configuring Multicast Routing (Required)
- Configuring the Virtual Interface (Required)
- Configuring VoIP Dial Peers (Required)
- Configuring E&M Voice Ports (Required, if used)
- Configuring for Receive Only Mode (Optional)
- Configuring Relevant Interface (Serial/Ethernet) (Required)
- Configuring Voice Ports in High-Density Voice Network Modules (Required, if using T1/E1)

Configuring Multicast Routing (Required)

To enable multicast routing on the platform, perform the following steps:

	Command	Purpose
Step 1	`Router(config)#` **`ip multicast-routing`**	Enables multicast routing.

Configuring the Virtual Interface (Required)

To configure the virtual interface for multicast fast switching, perform the following steps:

	Command	Purpose
Step 1	`Router(config)#` **`interface`** *`vif1`*	Defines 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 2	`Router(config-if)#` **`ip address`** *`address subnet-mask`*	Assigns the IP address and subnet mask for the virtual interface.
Step 3	`Router(config-if)#` **`ip pim`** { **`sparse mode`** | **`dense-mode`** | **`sparse-dense-mode`** }	Specifies Protocol Independent Multicast (PIM). Whatever mode you choose should match all the interfaces in all the routers of your network.

Configuring VoIP Dial Peers (Required)

To configure the VoIP dial peers on the router, perform the following steps:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`tag`* **`voip`**	Assigns a variable number (*tag*) to the VoIP dial peer.
Step 2	`Router(config-dial-peer)#` **`destination-pattern`** *`multicast-session-number`*	The destination pattern for the VoIP dial peer must match the value of the *multicast-session-number* string for the corresponding voice port.
Step 3	`Router(config-dial-peer)#` **`session protocol multicast`**	This step is mandatory for voice multicasting and is the command introduced specifically for the Cisco Hoot and Holler over IP application.
Step 4	`Router(config-dial-peer)#` **`session target ipv4:`***`address`*`:`*`port`*	Assigns 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. An odd-numbered port (UDP port number + 1) is used for the RTCP traffic for that session.

	Command	Purpose
Step 5	`Router(config-dial-peer)# ip precedence number`	(Optional) Specifies the IP precedence.
Step 6	`Router(config-dial-peer)# codec (codec-type}`	Configures the codec. You must configure the same codec on all dial peers in a session. When the default codec, **g729r8**, is used, it does not appear in the configuration when the **show running** command is used.

Configuring E&M Voice Ports (Required, if used)

If using E&M voice ports, configure them by performing the following steps:

	Command	Purpose
Step 1	`Router(config)#` **`voice class permanent`** *`tag1`*	Defines voice class for transmit-receive mode.
Step 2	`Router(config-class)#` **`signal timing oos timeout disabled`**	Disables signaling loss detection.
Step 3	`Router(config-class)#` **`signal keepalive`** *`number`*	Specifies keepalive signaling packet interval.
Step 4	`Router(config-class)#` **`exit`**	Returns to global configuration mode.
Step 5	`Router(config)#` **`voice-port`** `router-`*`slot/voice-slot/VIC-port`*	Selects the voice port to configure.
Step 6	`Router(config-voice-port)#` **`voice-class permanent`** *`tag1`*	Uses voice class *tag1* for the port that is allowed to speak.
Step 7	`Router(config-voice-port)#` **`connection trunk`** *`multicast-session-number`*	Ties the voice port to a multicast-session number.
Step 8	`Router(config-voice-port)#` **`music-threshold`** *`threshold`*	(Optional) Sets the music threshold to make VAD less sensitive.
Step 9	`Router(config-voice-port)#` **`operation 4-wire`**	Specifies 4-wire operation. (2-wire is the default.)
Step 10	`Router(config-voice-port)#` **`type`** `{` **`1`** `|` **`2`** `|` **`3`** `|` **`5`** `}`	Selects the appropriate E&M interface type (depending on the end connection—such as PBX): • Type 1 indicates the following lead configuration (default—this is the recommended option): – E—Output, relay to ground – M—Input, referenced to ground • Type 2 indicates the following lead configuration: – E—Output, relay to SG – M—Input, referenced to ground – SB—Feed for M, connected to -48V – SG—Return for E, galvanically isolated from ground • Type 3 indicates the following lead configuration: – E—Output, relay to ground – M—Input, referenced to ground – SB—Connected to -48V – SG—Connected to ground • Type 5 indicates the following lead configuration: – E—Output, relay to ground – M—Input, referenced to -48V

	Command	Purpose
Step 11	`Router(config-voice-port)# signal { wink-start	immediate	delay-dial }`	Configures the signaling type for E&M voice ports. The default is **wink-start**. Select **immediate** for the Cisco Hoot and Holler over IP application. In the immediate-start protocol, the originating side does not wait for a wink before sending addressing information. After receiving addressing digits, the terminating side then goes off-hook for the duration of the call. The originating endpoint maintains off-hook for the duration of the call.
Step 12	`Router(config-voice-port)# voice-port router-slot/voice-slot/VIC-port`	Selects another voice port.
Step 13	`Router(config-voice-port)# voice-class permanent tag2`	Uses voice class *tag2* for the receive-only port.
Step 14	`Router(config-voice-port)# connection trunk multicast-session-number`	Ties the voice port to the same multicast-session number as in Step 12.
Step 15	`Router(config-voice-port)# music-threshold threshold`	(Optional) Sets the music threshold to make VAD less sensitive.
Step 16	`Router(config-voice-port)# operation 4-wire`	Specifies 4-wire operation. (2-wire is the default.)

Configuring for Receive Only Mode (Optional)

To configure Cisco Hoot and Holler over IP as receive-only mode, perform the following steps:

	Command	Purpose
Step 1	`Router(config-class)# voice class permanent tag2`	Defines voice class for receive-only mode.
Step 2	`Router(config-class)# signal pattern oos receive 0000`	Specifies the received signal pattern.
Step 3	`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 4	`Router(config-class)# signal keepalive number`	Specifies keepalive signaling packet interval.

Configuring Relevant Interface (Serial/Ethernet) (Required)

To configure either the serial or Ethernet interface, perform the following steps:

	Command	Purpose
Step 1	`Router(config)# interface { ethernet	serial } slot/port`	Configures the physical interface (serial/Ethernet) for transmitting multicast packets.
Step 2	`Router(config-if)# ip address address subnet-mask`	Assigns the IP address and subnet mask for the interface.

	Command	Purpose
Step 3	`Router(config-if)#` **`ip pim`** `{` **`sparse mode`** `|` **`dense-mode`** `|` **`sparse-dense-mode`** `}`	Specifies Protocol Independent Multicast (PIM). Whatever mode you choose should match all the interfaces in all the routers of your network.
Step 4	`Router(config-if)#` **`no shutdown`**	Enables the interface.

Configuring Voice Ports in High-Density Voice Network Modules (Required, if using T1/E1)

A multiflex trunk interface card (NM-HDV) in a high-density voice network module requires special voice-port configuration when connecting for T1/E1 operation. Perform the following steps:

	Command	Purpose
Step 1	`Router(config)#` **`voice class permanent`** *`tag1`*	Defines voice class for transmit-receive mode.
Step 2	`Router(config-class)#` **`signal timing oos timeout disabled`**	Disables signaling loss detection.
Step 3	`Router(config-class)#` **`signal keepalive`** *`number`*	Specifies keepalive signaling packet interval.
Step 4	`Router(config-class)#` **`exit`**	Returns to global configuration mode.
Step 5	`Router(config)#` **`voice-card`** *`number`*	Selects the card to configure.
Step 6	`Router(config-voicecard)#` **`codec complexity high`**	Codec complexity must be high. Voice multicasting does not support medium complexity, which is the default.
Step 7	`Router(config)#` **`controller`** `{` **`t1`** `|` **`e1`** `}` *`slot/port`*	Selects the T1 or E1 controller to configure.
Step 8	`Router(config-controller)#` **`ds0-group`** *`ds0-group-number`* **`timeslots`** *`timeslot-list`* **`type`** *`type`*	Maps a group of time slots to a DS0 group.
Step 9	`Router(config)#` **`voice-port`** *`slot/port:ds0-group-number`*	Configures a DS0 group that was created in Step 4.
Step 10	`Router(config-voice-port)#` **`connection trunk`** *`multicast-address`*	Ties the connection trunk to a multicast address. This command is repeated for each DS0 group. All groups use the same multicast address, if connecting to the same multicast session.
Step 11	`Router(config-voice-port)#` **`voice-class permanent`** *`tag2`*	Uses voice class *tag2* for the receive-only port.

Configuration Examples

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

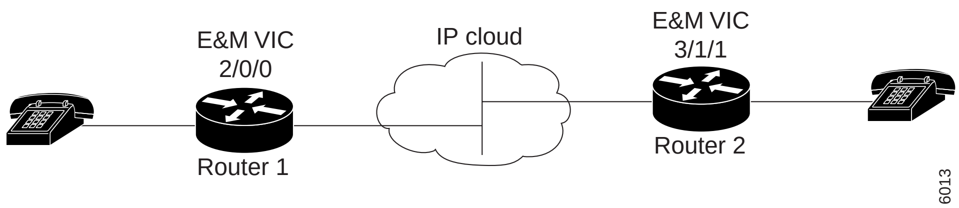
- Voice Multicasting over an Ethernet LAN, page 18
- Voice Multicasting over a WAN, page 22
- Cisco Hoot and Holler over IP with Ethernet Topology (Two Hoot Groups)
- Cisco Hoot and Holler over IP with Frame-Relay Topology (One Hoot Group)

Note In all of the following configuration examples, the routers are configured with an **interface vif1** command. This is a virtual interface that is similar to a loopback interface—it is a logical IP interface that is always up when the router is active. In addition, it must be configured so that the Cisco Hoot and Holler over IP packets that are locally mixed on the DSPs can be fast-switched along with the other data packets. This interface needs to reside on its own unique subnet, and that subnet should be included in the routing protocol updates (RIP, OSPF, and so on).

Voice Multicasting over an Ethernet LAN

Figure 4 shows the simplest configuration for voice multicasting over an Ethernet LAN. Two routers are connected to each other over the Ethernet LAN. One E&M phone is connected to each router.

Figure 4 Voice Multicasting over a LAN



In router Abbott (Figure 4), the phone is connected to voice port 2/0/0, using the *router-slot/voice-slot/VIC-port* numbering convention. This router is configured as in the following example:

```
hostname Abbott
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
interface Vif1
 ip address 1.1.1.1 255.0.0.0
 ip pim sparse-dense-mode
!
interface Ethernet0/0
 ip address 3.3.3.1 255.0.0.0
 ip pim sparse-dense-mode
!
ip route 2.0.0.0 255.0.0.0 3.3.3.2
!
voice-port 2/0/0
 voice-class permanent 1
 connection trunk 111
 operation 4-wire
!
dial-peer voice 1 voip
 destination-pattern 111
 session protocol multicast
 session target ipv4:237.111.0.111:22222
!
```

Note 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.

Note The E&M voice port must be configured for 4-wire operation.

Configuring the Second Router

In router Costello (Figure 4), 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 from that of the first router.
- The voice port and slot should match the router's hardware configuration.

```
hostname Costello
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
!
interface Vif1
 ip address 2.2.2.2 255.0.0.0
 ip pim sparse-dense-mode
!
interface Ethernet0/0
 ip address 3.3.3.2 255.0.0.0
 ip pim sparse-dense-mode
!
ip route 1.0.0.0 255.0.0.0 3.3.3.1
!
voice-port 3/1/1
 voice-class permanent 1
 timeouts wait-release 3
 connection trunk 222
 music-threshold -30
 operation 4-wire
!
dial-peer voice 1 voip
 destination-pattern 111
 session protocol multicast
 session target ipv4:237.111.0.111:22222
!
```

Note The multicast session for this port, shown in the **session target** command, must match the multicast session configured on the first router.

The codec configured for this dial peer must match the codec for the dial peer on the first router.

Both routers must be configured to use the same connection trunk and destination pattern.

Verifying the Configuration

If you have configured your routers by 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.

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 show how to 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
```

The following commands show how to configure the voice ports on the multiflex trunk interface card:

```
!
voice-port 6/0:1
 connection trunk 111
!
voice-port 6/0:2
 connection trunk 111
!
voice-port 6/0:3
 connection trunk 111
 …
voice-port 6/0:22
 connection trunk 111
!
voice-port 6/0:23
 connection trunk 111
```

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 instructing 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 show how to 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
!
```

Note The configuration for the **codec g711ulaw** in the above configuration is not necessary—the default codec of **g729r8** could be used (and it would not display for **show config**).

Tips

- 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 routers 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 routers in a session. An odd-numbered port (UDP port number + 1) is used for the RTCP traffic for that session.
- The following codec restrictions apply:
  - You must configure the same codec on all dial peers and routers in a session.
  - Only G.711, G.726, and G.729 codecs are supported.
  - When the default codec, G.729r8, is used, it does not appear in the configuration.
- Voice activity detection (VAD) is enabled by default. Cisco recommends that 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!
```

Voice Multicasting over a WAN

The configuration for voice multicasting sessions over IP on Frame Relay is the same as for the Ethernet LAN in the previous example. Configure the WAN interface on each router with the **ip address** and **ip pim sparse-dense-mode** commands as shown in the section, Voice Multicasting over an Ethernet LAN.

Quality of Service

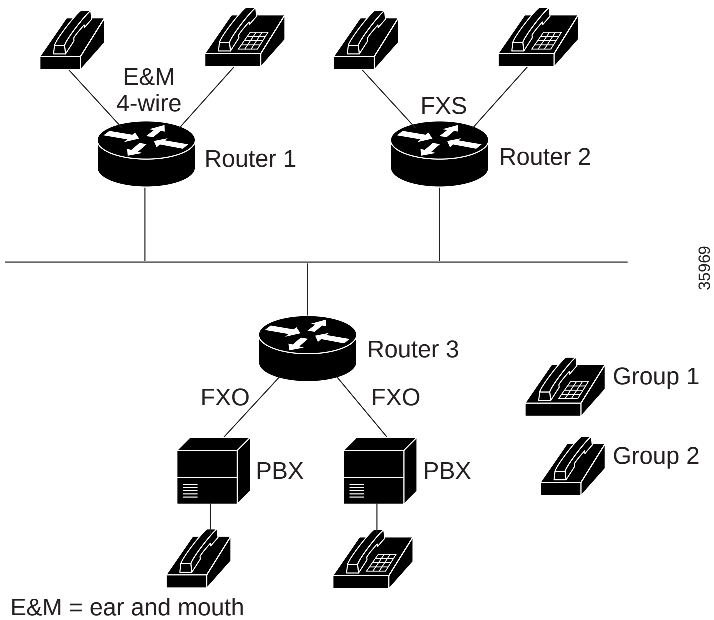
Voice traffic is much more sensitive to timing variations than data traffic. For good voice performance, configure your data network so that voice packets are not lost or delayed. The following example shows one way to improve quality of service (QoS) for voice multicasting over a Frame Relay connection:

```
!Configure physical interface for transmitting multicast packets.
!Listen to packets of Session Announcement Protocol (SAP).
!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.

Cisco Hoot and Holler over IP with Ethernet Topology (Two Hoot Groups)

Figure 5 Cisco Hoot and Holler over IP with Ethernet Topology



In this configuration, two hoot and holler groups are set up by defining two multicast groups (237.111.0.111 and 237.111.0.112) and mapping the **connection trunk** *111* and **connection trunk** *112* commands from the voice ports to the VoIP dial peers associated with each group. Each router is connected to a dedicated switch port, and IP precedence is set to 5 for all Cisco Hoot and Holler over IP packets.

Router-1 (E&M 4-Wire Ports)

```
hostname Router-1
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
interface Vif1
ip address 1.1.1.1 255.255.255.0
ip pim sparse-dense-mode
!
!
interface Ethernet0/0
ip address 1.5.13.1 255.255.255.0
ip pim sparse-dense-mode
!
router rip
network 1.1.1.0
network 1.5.13.0
!
voice-port 1/0/0
voice-class permanent 1
connection trunk 111
```

```
music-threshold -30
operation 4-wire
!
voice-port 1/0/1
voice-class permanent 1
connection trunk 112
music-threshold -30
operation 4-wire
!
dial-peer voice 111 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.111:22222
ip precedence 5
!
dial-peer voice 112 voip
destination-pattern 112
session protocol multicast
session target ipv4:239.194.0.10:22224
ip precedence 5
!
end
```

Router-2 (FXS Ports)

```
hostname Router-2
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
interface Vif1
ip address 2.2.2.2 255.255.255.0
ip pim sparse-dense-mode
!
interface Ethernet0/0
ip address 1.5.13.2 255.255.255.0
ip pim sparse-dense-mode
!
router rip
network 2.2.2.0
network 1.5.13.0
!
dial-peer voice 111 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.111:22222
ip precedence 5
!
dial-peer voice 112 voip
destination-pattern 112
session protocol multicast
session target ipv4:239.194.0.10:22224
ip precedence 5
!
end
```

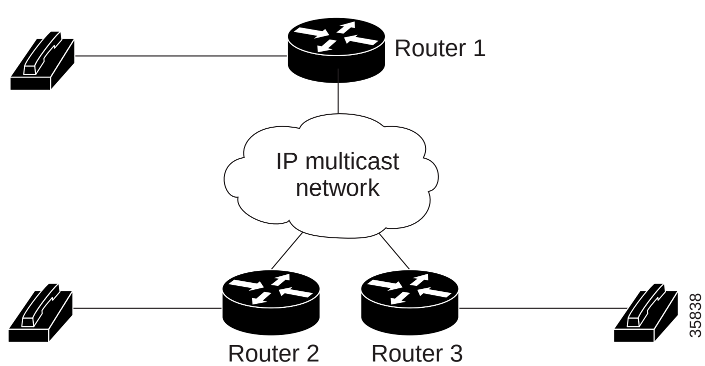
Note If you want to join the hoot and holler session directly without having to dial any session numbers, use the command **connection plar**, followed by the multicast-session number.

Router-3 (FXO Ports)

```
hostname Router-3
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
interface Vif1
ip address 3.3.3.3 255.255.255.0
ip pim sparse-dense-mode
!
interface Ethernet0/0
ip address 1.5.13.3 255.255.255.0
ip pim sparse-dense-mode
!
router rip
network 3.3.3.0
network 1.5.13.0
!
dial-peer voice 111 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.111:22222
ip precedence 5
!
dial-peer voice 112 voip
destination-pattern 112
session protocol multicast
session target ipv4:239.194.0.10:22224
ip precedence 5
!
end
```

Cisco Hoot and Holler over IP with Frame-Relay Topology (One Hoot Group)

Figure 6 Cisco Hoot and Holler over IP with Frame-Relay Topology



In this topology, three routers are connected using 64K Frame-Relay PVCs in a hub and spoke topology, with Router 1 being the hub. We have defined one hoot and holler group. All three routers have been configured to traffic-shape their data and voice on the WAN to CIR, and all three routers are using IP RTP priority to guarantee QoS for the Cisco Hoot and Holler over IP packets. In addition, the frame-relay broadcast-queue is disabled on the serial interfaces. This occurs because, by default, the broadcast-queue is only 40 packets deep and Cisco Hoot and Holler over IP transmits packets at 50 packets per second. Unless the queue is disabled, some packets would be dropped and voice QoS would be degraded.

Router-1

```
hostname Router-1
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
interface Vif1
ip address 1.1.1.1 255.255.255.0
ip pim sparse-dense-mode
!
router rip
network 1.1.1.0
network 5.5.5.0
network 5.5.6.0
!
interface Serial0/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay
 frame-relay traffic-shaping
 no frame-relay broadcast-queue
!
interface Serial0/0.1 point-to-point
 ip address 5.5.5.1 255.255.255.0
 ip pim sparse-dense-mode
 frame-relay class hoot-n-holler
 frame-relay interface-dlci 100
 frame-relay ip rtp header-compression
!
interface Serial0/0.2 point-to-point
 ip address 5.5.6.1 255.255.255.0
 ip pim sparse-dense-mode
 frame-relay class hoot-n-holler
 frame-relay interface-dlci 101
 frame-relay ip rtp header-compression
!
map-class frame-relay hoot-n-holler
 frame-relay cir 128000
 frame-relay bc 1280
 frame-relay mincir 128000
 frame-relay fragment 160
 frame-relay ip rtp priority 16384 16384 128
 no frame-relay adaptive-shaping
!
voice-port 1/0/0
voice-class permanent 1
connection trunk 111
music-threshold -30
operation 4-wire
!
dial-peer voice 1 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.0:22222
```

```
ip precedence 5
```

Router-2

```
hostname Router-2
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
!
interface Vif1
ip address 2.2.2.2 255.255.255.0
ip pim sparse-dense-mode
!
router rip
network 2.2.2.0
network 5.5.5.0
!
interface Serial0/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay
 frame-relay traffic-shaping
 no frame-relay broadcast-queue
!
interface Serial0/0.1 point-to-point
 ip address 5.5.5.2 255.255.255.0
 ip pim sparse-dense-mode
 frame-relay class hoot-n-holler
 frame-relay interface-dlci 100
 frame-relay ip rtp header-compression
!
map-class frame-relay hoot-n-holler
 frame-relay cir 128000
 frame-relay bc 1280
 frame-relay mincir 128000
 frame-relay fragment 160
 frame-relay ip rtp priority 16384 16383 128
 no frame-relay adaptive-shaping
!
voice-port 1/0/0
voice-class permanent 1
connection trunk 111
music-threshold -30
operation 4-wire
!
dial-peer voice 1 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.0:22222
ip precedence 5
```

Router-3

```
hostname Router-3
!
ip multicast-routing
!
voice class permanent 1
signal timing oos timeout disabled
signal keepalive 65535
```

```
!
interface Vif1
ip address 3.3.3.3 255.255.255.0
ip pim sparse-dense-mode
!
router rip
network 3.3.3.0
network 5.5.6.0
!
interface Serial0/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay
 frame-relay traffic-shaping
 no frame-relay broadcast queue
!
interface Serial0/0.1 point-to-point
 ip address 5.5.6.2 255.255.255.0
 ip pim sparse-dense-mode
 frame-relay class hoot-n-holler
 frame-relay interface-dlci 101
 frame-relay ip rtp header-compression
!
map-class frame-relay hoot-n-holler
 frame-relay cir 128000
 frame-relay bc 1280
 frame-relay mincir 128000
 frame-relay fragment 160
 frame-relay ip rtp priority 16384 16383 128
 no frame-relay adaptive-shaping
!
voice-port 1/0/0
voice-class permanent 1
connection trunk 111
music-threshold -30
operation 4-wire
!
dial-peer voice 1 voip
destination-pattern 111
session protocol multicast
session target ipv4:237.111.0.0:22222
ip precedence 5
```

Command Reference

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

- **session protocol multicast**

session protocol multicast

To set the session protocol as multicast, use the **session protocol multicast** dial-peer configuration command. To negate this command and return to the *cisco* default session protocol, use the **no** version of this command.

session protocol multicast

no session protocol multicast

Syntax Description

There are no keywords or arguments.

Defaults

When this command is not implemented, the default session protocol is **cisco**.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced on Cisco 2600 and Cisco 3600 series routers for the Cisco Hoot and Holler over IP application.

Usage Guidelines

Use the **session protocol multicast** dial-peer configuration command for voice conferencing in a Hoot and Holler networking implementation. This command allows more than two ports to join the same session simultaneously. It is supported on Cisco 2600 and Cisco 3600 series routers.

Examples

The following example shows the use of the the **session protocol multicast** dial-peer configuration command in context with its accompanying commands:

```
Router(config)# dial-peer voice 111 voip
Router(config-dial-peer)# destination-pattern 111
Router(config-dial-peer)# session protocol multicast
Router(config-dial-peer)# session target ipv4:237.111.0.111:22222
Router(config-dial-peer)# ip precedence 5
Router(config-dial-peer)# codec g711ulaw
```

Related Commands

Command	Description
session target ipv4	Assigns the session target for voice-multicasting dial peers.

Glossary

CIR—Committed information rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame-Relay PVC.

Codec—Coder-decoder. Device that typically uses pulse-code modulation (PCM) to transform analog signals into a digital bit stream and digital signals back into analog signals. In Voice over IP, it specifies the voice-coder rate of speech for a dial peer.

Dial peer—An addressable call endpoint that contains configuration information including voice protocol, a codec type, and a telephone number associated with the call. There are five kinds of dial peers: POTS, VoIP, VoFR, VoATM, and VoHDLC. In Voice over IP, there are two kinds of dial peers:

- POTS—Connected through a traditional telephony network and points to a particular voice port on a voice network device.
- VoIP—Connected through a packet network (IP network for VoIP) and points to specific VoIP device.

DSP—Digital signal processor.

DTMF—Dual tone multifrequency. Uses two simultaneous voice-band tones for dial such as touch-tone.

E&M—Ear and mouth. Stands for the 2-wire or 4-wire interface with separate signaling paths for the receiving and transmitting signals—a type of signaling traditionally used in the telecommunications industry. Indicates the use of a handset that corresponds to the ear (receiving) and mouth (transmitting) component of a telephone. E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connecting. The Cisco analog E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines). E&M is also available on E1 and T1 digital interfaces.

FXO—Foreign Exchange Office. An FXO interface connects to the Public Switched Telephone Network (PSTN) central office and is the interface offered on a standard telephone. The Cisco FXO interface is an RJ-11 connector that allows an analog connection at the PSTN's central office or to a station interface on a PBX.

FXS—Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Cisco's FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

Hoot and holler—A broadcast audio network used extensively by the brokerage industry for market updates and trading. Similar networks are used in publishing, transportation, power plants, and manufacturing.

IVR—Interactive voice response. When someone dials in, IVR responds with a prompt to get a personal identification number (PIN), and so on.

PBX—Private branch exchange. Digital or analog telephone switchboard or switching facility located on the subscriber premises and used to connect private and public telephone networks.

PVC—Permanent virtual circuit.

QoS—Quality of service. QoS refers to the measure of service quality provided to the user.

TDM—Time-division multiplexing.

Trunk—Service that allows quasi-transparent connections between two PBXs, a PBX and a local extension, or some other combination of telephony interfaces to be permanently conferenced together by the session application and signaling passed transparently through the IP network.

VAD—Voice activity detection.

VIC—Voice interface card.

VNM—Voice network module.

VoIP—Voice over Internet Protocol.